THE MIXING ENGINEER'S HANDBOOK

BOBBY OWSINSKI

FIFTH EDITION

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The Mixing Engineer's Handbook 5th edition

by Bobby Owsinski

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Introduction

Welcome to the fifth edition of The Mixing Engineer's Handbook. There are a lot of changes and updates in this edition as I've done my best to adapt it to the latest trends in the world of mixing.

In the 20 years since I wrote the original book, the recording industry and mixing itself has undergone a huge shift. While the First and Second Editions were clearly in the age of the commercial studio centered around huge recording consoles, Edition Three saw the shift towards home recording. Edition Four saw mixers clearly mixing in-the-box (meaning in a digital audio workstation app), and now with Edition Five we see the home studio has fully matured with worldwide hits being made even on simple recording setups in a bedroom.

Not only that, plugin developers have now begun to leave the analog hardware emulations behind with new thinking about how audio can be processed and manipulated in digital domain. We also find audio plugins and apps are becoming more useful by taking much of the grunt work out of mixing, as many are now infused with artificial intelligence and machine learning.

All the more reason to update this book. Mixing techniques evolved and adapted to the digital world quite a while ago now, but with fewer studios, there are also fewer mentors to learn from. That said, the classic mixing techniques are still more useful than ever, since the basics of balance, equalization, compression, and effects never go out of

style regardless of the gear, music genre or release format that they're used on.

My main goal in writing the Mixing Engineer's Handbook has always been to preserve the classic and modern techniques before they're lost to rumor or twisted into irrelevance. Where once these skills were handed down from engineer to assistant, that whole master-apprentice information exchange has almost faded into oblivion, which is all the more reason to have a single repository of techniques.

For the fifth edition, I've added a number of new sections and interviews, and generally adapted the remaining material so that what's contained herein is much more relevant to today's DAW-based mixing. Since the majority of readers will be working at home in their personal studio, I've put a special emphasis on how the pros currently mix using their DAWs, as well as how to adapt their large-console techniques to the home studio.

Just so you know, the reason why I originally wrote the first edition of this book is probably the same reason why you're reading it: to get better at what I do. I noticed that my mixes were somewhat hit or miss. Sometimes they were great, sometimes just okay, and sometimes plain off the mark. I also noticed that much of the time my mixes didn't have the major label sound that I heard on the radio. Like you, I wanted this sound badly.

I was lucky enough that I live in Los Angeles and was already friends with some of the greatest mixers on the planet. I asked them to share their mixing secrets and they did without hesitation.

While doing research for the original version of this book, I found that a common factor among most great mixers was that they usually all had at least one mentor as a result of coming up through the studio ranks. Most of the mixers of my era (admittedly considered "classic" now) started as assistants, learned by watching and listening to the greats they helped, and had taken a little from all of them as a result.

I didn't do that, however. Being a musician first and foremost, I learned to engineer thanks to my early interests in electronics, which came from wanting to know how the electrons got from my guitar to the speakers of my amplifier. As I became familiar with the recording studio, I was lucky to be offered all sorts of varied engineering work, from recording jingles to big band to jazz to R&B to hard rock, but since I never wanted to give up being a musician (which I knew I'd have to do), I never took a proper studio job as an assistant to really learn the trade at the hands of the masters. As a result, my recording skills were always pretty good, but my mixing skills were lacking.

I soon realized that there were many others like me who were good but not great, not because they weren't capable, but because they didn't have the opportunity or access to the methods of the masters. After all, how often does a George Massenburg or Bruce Swedien record in Lincoln, Peoria, Santa Fe, or even smaller towns like Minersville, Millersburg, or Avondale? And unfortunately, because there are fewer real commercial studios left, there's even less of a chance of that information exchange happening today than ever before. Not only that, the vast majority of musicians (who inevitably end up as engineers in some capacity) operate from their personal studios anyway.

So the first edition of the book started out very selfishly, as it was meant specifically to meet my needs, but it ended up being for you as well. I hope you will benefit from it as much as I have.

And yes, my mixes have gotten much, much better.

When I wrote the first edition of The Mixing Engineer's Handbook, my intention was to interview as many great engineers as I could in order to accumulate their various methods and anecdotes simply as background material. The more I got into it, though, the more it became obvious that these interviews were living and breathing on their own and they really needed to be included in the text; otherwise, a lot of really useful information would be left out. In other words, let them tell you what they do in their own words.

These interviews are contained in Part II of the book. Many of the mixers interviewed in the previous four editions have been re-interviewed, since their mixing methods have changed along with the industry changes. Many started on a console but are now are totally "in-the-box."

Every one of the mixers I interviewed for this book was extremely forthcoming, answering just about any question and offering explicit information as to why and how he or she works. Professional jealousy just does not exist in this industry, at least in my experience, as the general attitude is, "I'll tell you anything you want to know, since no one can do it like me anyway."

As a matter of fact, here's a list of the engineers who contributed to this book, along with some of their credits. One big change in this edition. I've moved many of the interviews that appeared in the previous editions over to my website. You can access the full interviews (not just the edited ones that made the book) at: BobbyOwsinski.com/interviews

I've tried to include a mixer that represents every genre of modern music (punk to classic to alternative to jazz to classical to R&B to EDM to Latin to rap to orchestral to country), so there's something for everyone. I'll be quoting them from time to time, so I wanted to introduce them early on so you have some idea of their background when they pop up.

Just remember, whenever a "mixer" or "engineer" is referred to in this book, I don't mean your average, run-of-the-mill engineer (hardworking and well meaning as he or she is). I mean someone who's made the hits that you've listened to and loved. This book is about how these glorious few think, how they work, how they listen, and why they do the things they do. And even though we can't hear as they hear, perhaps we can hear through their words. Here are the mixers.

Bob Brockman (interview appears later in this book): "Bassy" Bob has been a fixture on the New York studio scene with a wide range of awards and credits that include Mary J. Blige, Toni Braxton, Notorious B.I.G., Babyface, Aretha Franklin, Al Green, the O'Jays, Brian McKnight, Jodeci, Faith Hill, Korn, Laurie Anderson, Vanessa Williams, Christina Aguilera, P. Diddy, Herbie Hancock, the Fugees, Santana, and Sting.

Bob Bullock (interview appears later in this book: Since he moved to Nashville in 1984, Bob has been one of the town's top engineers, trusted by the likes of Kenny Chesney, Shania Twain, George Strait, Reba McEntire, Hank Williams Jr., and Jimmy Buffett, among with many others. Prior to that he saw a different side of the music world while working in Los Angeles with acts such as the Tubes, Art Garfunkel, Seals and Crofts, Chick Corea, and REO Speedwagon.

Joe Chiccarelli (interview appears later in this book): With credits such as the White Stripes, Alanis Morissette, the Strokes, Jason Mraz, Tori Amos, Etta James, Beck, U2, Elton John, Oingo Boingo, the Shins, Frank Zappa, the Killers, Brian Setzer, and many more, chances are you've heard Joe's work more times than you know.

Richard Chycki (interview appears later in this book): Richard has worked with everyone from Rush to Dream Theater to Aerosmith to Mick Jagger and more. In fact, he has 26 gold and platinum records from Rush alone! Later in the book we're going to discuss one of the largest and most complex music projects ever.

Lee DeCarlo (interview appears on the interview site page): From his days as chief engineer at New York's Record Plant in the heady 1970s, Lee has put his definitive stamp on hit records that include works by Aerosmith, John Lennon, Kenny Loggins, Black Sabbath, Rancid, and Zakk Wylde, among many others.

Billy Decker (interview appears later in this book): Billy Decker has mixed hits for Kenny Chesney, Darius Rucker, Jason Aldean. Jaime Lynne Spears, George Jones and Sam Hunt, among others, and combined they've sold over 25 million albums. Billy is different from most other mixers in that he's able to mix extremely fast thanks to his templates that outline all the basic parameters needed to make each vocal and instrument work in the mix.

DJ Swivel (interview appears later in this book): Jordan Young, known to many as DJ Swivel, is a Grammy-winning producer, mixer and

songwriter who's worked with with a wide variety of hit-making artists that include Jay-Z, Diddy, Pharrell, Britney Spears, Beyonce, The Chainsmokers, BTS and many more. Jordan also has developed his own line of plugins based on his own unique processing techniques.

Jimmy Douglass (interview appears later in this book): One of the few engineers who can cross genres with both total ease and credibility, Jimmy has done records for artists as varied as Snoop Dogg, Jay-Z, the Roots, Ludacris, Justin Timberlake, Timbaland, Missy Elliott, Otis Redding, The Rolling Stones, Foreigner, Hall & Oates, Roxy Music, and Rob Thomas.

Benny Faccone (interview appears later in this book): Benny is unique in that he's a Canadian from Montreal, but 99 percent of the projects he works on are Spanish. From Luis Miguel to Ricky Martin to the Latin rock supergroup Mana, to Spanish remixes for Boys II Men, Tony Braxton, and Sting, Benny's 18-time Grammy-winning work is heard far and wide around the Latin world.

Jerry Finn (interview appears appears on the interview site page): With credits from Green Day to Rancid to the Goo Goo Dolls to Beck, Jerry represented one of the new generation of mixers who knows all the rules but are perfectly willing to break them. Unfortunately, Jerry passed away in 2008, but his techniques and wisdom live on.

Jon Gass (interview appears later in this book): Jon has long been the go-to mixer for a who's who of music superstars, including Madonna, Whitney Houston, Janet Jackson, Celine Dion, Mariah Carey, Mary J. Blige, Usher, Babyface, Earth, Wind & Fire, Lionel Richie, John Mellencamp, and many more.

Don Hahn (interview appears on the interview site page): When it comes to recording and mixing a 45- to 100-piece orchestra, there was no one better than Don, with an unbelievable list of credits that range from major television series to such legends as Count Basie, Barbra Streisand, Chet Atkins, Frank Sinatra, Herb Alpert, Woody Herman, Dionne Warwick, and a host of others (actually, 10 pages more).

Andy Johns (interview appears on the interview site page): Andy Johns needs no introduction because we've been listening to the music that he's been involved with for most of our lives. With credits such as Led Zeppelin, Free, Traffic, Blind Faith, The Rolling Stones, and Van Halen (to name just a few), Andy has set a standard that most mixers are still trying to live up to.

Bernie Kirsh (interview appears on the interview site page): From virtually all of Chick Corea's recordings to Quincy Jones, Stanley Clarke, Joe DeFrancesco, and Al Di Meola, Bernie has certainly made his mark as one of the top engineers in the world of jazz.

Dan Korneff (interview appears later in this book): Producer, mixer and engineer Dan Korneff has not only worked with prominent names like Breaking Benjamin, Paramore, Papa Roach, Lamb of God and My Chemical Romance, but has developed some of the coolest audio plugins available through his company Korneff Audio.

Nathaniel Kunkel (interview appears on the interview site page): One of the most in-demand mixers in the business, with credits that range from James Taylor, Lionel Richie, and Sting to Good Charlotte, Fuel,

and Insane Clown Posse, Nathaniel represents the best of the next generation of mixers.

George Massenburg (interview appears later in this book): From designing the industry's most heralded audio tools to engineering classics by Little Feat, Earth, Wind & Fire, Dixie Chicks, James Taylor, Billy Joel, Lyle Lovett, and Linda Ronstadt (to name only a few), George needs no introduction to anyone even remotely connected to the music or audio business.

Andrew Maury (interview appears later in this book): After starting his career in 2008 mixing front-of-house with Ra Ra Riot while producing and mixing small bands and artists between tours, Andrew has since gone on to mix projects for Shawn Mendez, Post Malone, Lizzo, Kimbra, and many more.

Greg Penny (interview appears later in this book): Born into a music-business family to bandleader/producer Hank Penny and hit recording artist Sue Thompson, Surround Music Award winner Greg Penny seemed destined for a life in the studio. Indeed Greg's production aspirations resulted in hits with k.d. lang, Cher, and Paul Young among others, but a meeting with Elton John while in his teens turned into an award-winning mixing journey with the legend many years down the road.

David Pensado (interview appears later in this book): Over the last two decades, Dave has taken mixing to a new level in artistry, having mixed megahits for superstars such as Christina Aguilera, Justin Timberlake, Kelly Clarkson, Pink, Black Eyed Peas, Beyonce, Shakira, and Michael Jackson, among many others. Well known in the business way before

his popular Pensado's Place web video series, Dave not only is on the cutting edge of technology, but has thought long and hard about the more cerebral aspects of mixing as well.

Elliot Scheiner (interview appears later in this book): With a shelf full of industry awards (seven Grammys, an Emmy, four Surround Music Awards, the Surround Pioneer and Tech Awards Hall of Fame, and too many total award nominations to count) from his work with The Eagles, Steely Dan, Fleetwood Mac, Sting, John Fogerty, Van Morrison, Toto, Queen, Faith Hill, Lenny Kravitz, Natalie Cole, Beyonce, the Doobie Brothers, Aerosmith, Phil Collins, Aretha Franklin, Barbra Streisand, and many, many others, Elliot has long been widely recognized for his artful and pristine mixes.

Andrew Scheps (interview appears later in this book): Andrew Scheps has brought a perfect combination of old- and new-school skills to his work with a who's who of superstar artists, including Red Hot Chili Peppers, Metallica, U2, Justin Timberlake, Jay-Z, The Rolling Stones, Linkin Park, Jewel, Neil Diamond, and Adele.

Ken Scott (interview appears later in this book): Legendary producer/engineer Ken Scott began his career working with The Beatles on The White Album and Magical Mystery Tour; on six David Bowie records, including the seminal Ziggy Stardust album; and with Pink Floyd, Elton John, Duran Duran, Jeff Beck, Supertramp, Procol Harum, Devo, Kansas, Mahavishnu Orchestra, and many more. To put it mildly, he's an absolute icon in the recording industry, having been a part of records that have conservatively sold more than 200 million units.

Ed Seay (interview appears later in this book): Ed has become one of the most respected engineers in Nashville since moving there in 1984, helping to mold hits for major hit-makers such as Blake Shelton, Lee Brice, Martina McBride, Ricky Skaggs, Dolly Parton, Pam Tillis, Highway 101, Collin Raye, and a host of others.

Allen Sides (interview appears on the interview site page): Although well known as the owner of the premier Ocean Way Studio complex in Los Angeles, Allen is one of the most respected engineers in the business, with credits that include Josh Groban, Michael Jackson, Chris Botti, Barry Manilow, Neil Diamond, Mary J. Blige, and Faith Hill, as well as many major film scores.

Don Smith (interview appears on the interview site page): With credits that read like a who's who of rock and roll, Don has lent his unique expertise to projects by The Rolling Stones, Tom Petty, U2, Stevie Nicks, Bob Dylan, Talking Heads, The Eurythmics, The Traveling Wilburys, Roy Orbison, and Iggy Pop, among many more. Don is another who unfortunately passed away way too soon, but hopefully this book will keep his unique techniques alive.

Ed Stasium (interview appears on the interview site page): Ed has made some great guitar albums like the ones by The Ramones, The Smithereens, and Living Color, but he has also worked with the likes of Mick Jagger, Talking Heads, Soul Asylum, Motorhead, and even Gladys Knight and the Pips and Ben Vereen.

Bruce Swedien (interview appears on the interview site page): Maybe the most revered of all engineers, Bruce has a credit list that could take up a chapter of this book alone. Although the biggest Michael Jackson

albums (Off the Wall, Thriller, Bad, Dangerous) would be enough for most mixer's resumes, Bruce can also include such legends as Count Basie, Tommy Dorsey, Duke Ellington, Woody Herman, Oscar Peterson, Nat "King" Cole, George Benson, Mick Jagger, Paul McCartney, Patti Austin, Edgar Winter, and Jackie Wilson, among many, many others.

For those of you who don't have the time or desire to read each interview, I've summarized many of their working methods in Part I of the book.

Remember, some of these interviews are located at BobbyOwsinski.com/interviews and are well-worth checking out.

Please note: Just because you read this book doesn't automatically guarantee that you'll become a platinum-selling mixer who makes lots of money and works with big-name recording artists. You'll get many tips, techniques, and tricks from the book, but you still need ears and experience, which only you can provide. All this book can do is point you in the right direction and help a little on the way!

Also keep in mind that just because one best-selling mixer might do things a certain way or use a certain technique or plugin, that doesn't mean that's the only way to do it. In fact, you'll notice that what works for one mixer may be completely opposite of what works for another, yet they both produce great mixes. Plus, mixers tend to change the way they work over their careers, so there's a good possibility that if you come back in fives years from now they may have a different approach to certain aspects of their mixes.

You should always feel free to experiment, because, after all, whatever works for you is in fact the right way.

PART I

Mixing Techniques

1

Some Background

Before we get into the actual mechanics of mixing, it's important to have some perspective on how this engineering skill has developed over the years.

The Evolution of Mixing

It's obvious to just about everyone who's been around long enough that mixing has changed a lot over the decades, but the whys and how's aren't quite so obvious.

Back in the 1950s, the early days of what we'd consider modern sound recording, mixing was minimal at best because the recording was made with a single-track mono tape machine and a big recording session meant that all four of the studio's microphones (if they were lucky to have that many) were used. If more than one microphone was used, they were fed into a 2 to 4 channel mixer intended for radio or television broadcast because that's all that was available at the time. The primary goal at this point was to capture a live performance.

Over the years recording evolved from capturing an unaltered musical event to one that was artificially created through overdubs, thanks to the 1955 introduction of Sel-Sync (the ability to play back from the tape machine's record head so everything stayed in sync). The availability of more and more tracks from a tape machine led to larger and larger

consoles, starting with the Studer Model 69 in 1958, the first mixer created mainly for studio recording.

Though there may have been some overdubs, most recordings still revolved around capturing a live performance in the studio. Revered engineer/producer Eddie Kramer (engineer for Jimi Hendrix, Led Zeppelin, KISS, and many rock superstars) shares his experience from those early days of recording particularly well:

"Everything was 4-track [when I started recording], so we approached recording from a much different perspective than people do nowadays. My training in England was fortunately with some of the greatest engineers of the day who were basically classically trained in the sense that they could go out and record a symphony orchestra and then come back to the studio and do a jazz or pop session, which is exactly what we used to do. When I was training under Bob Auger, who was the senior engineer at Pye Studios, he and I used to go out and do classical albums with a 3-track Ampex machine and three Neumann U47s and a single three-channel mixer. With that sort of training and technique under my belt, approaching a rock-n-roll session was approaching it from a classical engineering standpoint by making the sound of a rock band bigger and better than it was. But the fact of the matter was that we had very few tools at our disposal except EQ, compression, and tape delay. That was it."

English mixer Andy Johns, who apprenticed under Kramer and eventually went on to equally impressive credits with the Rolling Stones, Led Zeppelin, Traffic, Van Halen, and others, goes a step further.

"You know why [The Beatles seminal album] Sgt. Pepper sounds so good? You know why [The Jimi Hendrix Experience] Are You Experienced sounds so good, almost better than what we can do now?

Because when you were doing the 4 to 4 [mixing down from one four-track machine to another to open up additional tracks for recording], you mixed as you went along. There was a mix on two tracks of the second 4-track machine, and you filled up the open tracks and did the same thing again. You mixed as you went along; therefore, after you got the sounds that would fit with each other, all you had to do was adjust the melodies.

Nowadays, because you have this luxury of the computer and virtually as many tracks as you want, you don't think that way anymore."

As Andy Johns alludes to, eventually recording consoles took a major leap in technology. As more and more tape tracks became the norm for music recording (eventually leading to 24 and 48 track sessions), computer automation and parameter recall soon became a required feature on all consoles just to manage the complexity of these far larger sessions. With all that came not only an inevitable change in the philosophy of mixing, but even a change in the way that a mixer listened or thought as well.

And indeed, once more tracks were available and many elements began to be recorded in stereo (and now sometimes even in multichannel surround), the emphasis has turned from the bass anchoring the record to the big beat of the drums as the main focal point. This is partially because typical drum miking went from just overhead and kick-drum mics to the now-common occurrence of a mic on every drum, since the

consoles now incorporated more microphone inputs and there were plenty of tracks to record on.

Since the drums could be spread out over 6 or 8 or even 20 tracks, they could now be listened to more carefully during the mix, since they didn't have to be premixed along with the bass onto only one or two tracks. Instead of the drums being thought of as just another instrument equal to the bass, now they demanded more attention because more tracks were used.

At that time (approximately 1975), thanks to the widespread use of the then-standard 24-track tape machine, mixing changed forever, and, for better or for worse, began to evolve into what it is today.

The Need To Automate

With more tracks came more mix complexity. Early multitrack mixing became a multi-man operation, with four or five sets of hands on different console parameters, every one being given a job to execute during the mix. These sessions became a communal performance by the engineer, producer, assistant, and band members. Of course, it was impossible to execute the mix perfectly all the way through with that much human interaction, so mixes became a collection of sections edited together, but they were holistic and organic and part of the charm of the late '60s- and '70s-era records.

The demand for more mixing precision brought about console automation, first affecting only the console channel faders and mutes.

Now it was possible to reduce the number of humans involved with a mix, since only the console parameters such as EQ and effects sends required manual dexterity.

Soon the demands for remixes from record-label executives required that the massive outboard gear setups that became the norm had to be rebuilt in order to update mixes. The mixing performance had to be recreated even though the only change requested was only the vocal level be raised by a dB in the choruses. This brought about the need for "Total Recall," the feature that made SSL consoles a must-have for every major studio.

While it was now possible to manually recall every position of every parameter on a console, assistants still had to fill in elaborate sheets to manually recall every piece of outboard gear as well. A typical recall setup could take three or four hours alone before the first notes of the song even came out of the speakers.

The next innovation was "resettable" consoles that would not only remember all of your parameter settings, but would automatically reset them so the assistant didn't have to do it manually. As consoles became larger and larger to the point where 56 channels was soon considered small, this was almost a necessity. Of course, the outboard gear still had to be reconnected and reset by hand, and with some mixers using 20, 30, or even more pieces during a large mix, life in the studio became more and more complex.

But an interesting turning point occurred around 2001. With the computer-based digital audio workstation (DAW) now becoming more and more the centerpiece of the studio, much of the automation and

effects began to take place inside the DAW application ("in the box" became the commonly used phrase), eliminating the need for much of the outboard gear used on every mix. Soon mixers became more comfortable with the sound of mixing completely inside the DAW, and thanks to the wide variety of digital controllers available that supplied faders and knobs, they had the same tactile experience as in the analog world of consoles.

Mixing in the box had another big effect on the music business though. With album-project budgets dropping to the point where they almost matched the price of buying a full DAW setup, many mixers were suddenly faced with the scenario of "We can either pay for you or for the studio, but not both." This forced many top mixers to move their base of operations from a commercial studio into a studio inside their homes.

Today even free or low-cost DAW applications are far more powerful than what major acts were used to using from the '50s through the '80s. It's an amazing time to be an engineer, if you have a handle on what the tools at your disposal are able to accomplish.

Of course music has changed a lot as well, since so much is now created in a workstation using samples and loops, with just a few musical overdubs and vocals. This has also led to more streamlined sessions. While it's hardly worth mentioning a 100+ track session these days, most of these tracks may be just alternate takes, as many songs have fewer final mix elements than ever. As music evolves, so does the technology with it, although we've seen over the years that many times it's the tech that leads the way.

Different Mixing Styles

Once upon a time, engineers worked for one particular studio, and one of the reasons a client would book time there was to get the services of that particular engineer. Because the engineer was tied to a specific region of the world, a unique mixing style for the area developed (much like what happened with the music), thanks to engineers, producers and artists exchanging tips and tricks with one another. As a result, until the late 1980s or so, it was easy to tell where a record was made just by the sound of the mix.

Today there’s less of a distinction than there used to be between the mixing approach of different areas. The homogenization of styles first came from the fact that now-independent engineers mixed in a variety of locations around the world, with many having relocated to new areas, transplanting their mixing styles along the way. And the techniques that used to be so regional are now freely and easily spread around the world thanks to courses like my Top 40 Mixing Secrets and video platforms like YouTube.

That said, the mixing styles of today can be traced to four major styles from the past, where most mixes took place: New York, Los Angeles, London, and Nashville. If you listen to records from the ’80s and ’90s, you can distinctly hear each one.

The New York Style

The New York style used to be perhaps the easiest to identify because it featured a lot of compression, which makes the mix very punchy and aggressive (just like New Yorkers). In many cases, the compressed instruments (mostly the rhythm section) are even recompressed several times along the way.

It seems that every New York engineer that I've ever talked to (even the transplanted ones) used the same trick, which is to send the drums (sometimes with the bass) into a couple of busses, send that through some compressors, squeeze to taste, then add back a judicious amount of this compressed rhythm section to the mix through a couple of channels.

This effect can be enhanced even further by boosting the high and low frequencies (lots of boost in many cases) to the compressed signal as well. (More on this "New York Compression Trick" later in the book in Chapter 9, The Dynamics Element: Compression, Limiting, Gating, and De-Essing.)

The LA Style

The LA style exhibited a somewhat more natural sound, which, although also compressed, is done to a less obvious degree than the New York style. There's also less effects layering than the London style, but a good bit of delayed reverb is added.

The LA style always tried to capture a musical event and then sonically augment it, rather than recreate it. Some good examples would be any

of the Doobie Brothers or Van Halen hits of the ‘70s and ‘80s.

The London Style

The London sound was a highly layered musical event that borrowed from the New York style in that it would be pretty compressed but had multiple effects layers that put each mix element into its own distinct sonic environment. Although the musical arrangement is important to any good mix, it’s even more of a distinctive characteristic of a London mix.

What this means is that many mix elements appear at different times during a mix, some for effect and some to change the dynamics of the song. Each new element would be in its own environment and, as a result, would have a different ambient perspective. A perfect example of this would be Hugh Padgham’s work with the Police, or just about anything produced by Trevor Horn, such as Seal or Grace Jones or Yes’s “Owner of a Lonely Heart.”

The Nashville Style

Nashville has gone through various phases through the years where the mixing style has evolved. At one point in time, the songs were so dependent on the artist that the vocal sat way out in front of the music bed, sometimes almost to the point where they both seemed almost disconnected.

The Nashville style today has evolved (some might say devolved) from what it was during the ’60s and ’70s to become much more like the modern compressed version of the LA style of the ’70s. Says legendary Nashville engineer/producer Ed Seay:

“Back when I used to listen to my dad’s old Ray Price and Jim Reeves country records, they weren’t very far from what pop was in the early ’60s: very mellow, big vocals, very subdued band, very little drums, strings, horns, lush. Mix-wise, there wasn’t really too much difference in an Andy Williams record and one of the old Jim Reeves records.

What happened was that country got too soft-sounding. You’d cut your track and then do some sweetening with some horns and strings. At one time strings were on all the country records, and then it kind of transformed into where it’s at today, with almost no strings on country records except for big ballads. For the most part, horns are completely dead. They’re almost taboo. Basically it’s rhythm track–driven and not really very far off from where pop was in the mid to later ’70s. The Ronstadt “It’s So Easy to Fall in Love” and “You’re No Good,” where you hear guitar, bass, drums, keyboards, a slide or steel, and then a vocal background; that’s pretty much the format now, although fiddle is used also.

Ironically enough, a lot of those guys that were making those records have moved here because, at this point, this is one of the last bastions of live recording.”

Nowadays there’s far less difference between mixing styles than there was during the ’50s to the ‘80s, but variations still do exist. Although the style differences blur on most music, electronic dance music still has

considerable variation divided around the traditional geographic boundaries of New York, LA, and London, with additional pockets in major cities around the world.

Other Styles

Increased globalization has had its effect on regional styles. Philadelphia, Memphis, Ohio, Miami, Atlanta and San Francisco all had sub-styles of the Big Four, the globetrotting lifestyle of most A-list engineers in the ’90s caused a homogenization of these regional styles.

Where at one time most studios had house engineers, the market became predominately made up of freelancers that frequently traveled from studio to studio and project to project, bouncing between different cities (and therefore styles) as easily as flipping the channel on a TV.

Also, while an engineer might have changed studios but remain located in a specific area all his or her working life, it became commonplace for an in-demand engineer to relocate to several major media centers during the course of his career. Because of this movement, a cross-pollination of styles started to blur the distinction between the Big Four in the ‘90s.

Today the differences between mixing styles are minor as compared to the way they used to be. Now nearly everyone mixes in the box because of the quick turnaround in revisions, which wasn’t true in the heyday of analog. And for many veterans of a previous era when consoles

ruled, they’re getting results that many feel are about the same. Even if there is a difference, very few complain about it any more.

In truth, the major differences in mixing style came in around 2001 with the gradual acceptance of the DAW as the studio centerpiece. Thanks to the Internet and books like this, the styles are now more genre specific than regional.

12 Reasons Why Studio Mixing Is Different from Live Mixing

You may have a lot of experience mixing a live band, but mixing in the studio is a distinctively different experience. The thought process is different, the mindset is different, the approach is different and the chain of command is different.

In an effort to contrast these two different experiences, let’s move from the most evident differences to those that are, shall we say, a bit more subtle.

Repertoire. Most live gigs rarely change repertoire much from gig to gig. You can hone the mix for each song the more times you gig. In the studio, each song is unique and fresh, and when it’s finished, it’s on to the next one. This doesn’t apply to a house mixer though, since each set is a new and different experience.

Scrutiny. On a live gig, your mix is gone as soon as the song is over. In the studio, what you do is under a microscope and will likely be analyzed, dissected, and reorganized, all in the name of making the mix stronger.

Equipment. The gear you use on a live gig won't always translate to the studio. You choose the gear for a gig based upon versatility, durability, and general ruggedness. The only thing that counts in the studio is the sound. While one size might fit all on a gig in terms of gear such as compressors, delays, and reverbs, the presets that are frequently used usually make for a boring studio mix. And don't forget that the studio world is now one of plugins. The studio requires a wide range of sonic possibilities, so you'll need to have and know how to use a number of gear or plug-in choices to get there.

Leadership. On a gig you have a bandleader that makes the set list, counts off the songs, may direct the solos, and ends the songs, but how you mix is pretty much up to you. In the studio you're answering to a hierarchy consisting of the producer, the artist, management, and most likely, someone at the record label if the artist is signed. The producer is the final decision-maker, with ultimate authority over everything you do, although the artist has much say in the final product as well.

Nuance. The little things count in the studio. Everything you do can be critical to a mix, so nuances are just as important as the basic balance of the instruments (as you'll see all throughout this book). During a live gig, the nuances are usually gone in the wind, overcome by the stage volume, the acoustics, and the attention span of the players and audience. In the studio, everything you do is scrutinized because it's all captured. What that means is you've got to be at the top of your game on every song.

Etiquette. You might get away with being a jerk on a live gig since the band or others on the crew usually will put up with you (to a point) as long as you do your job well. Not so in the studio. If you make someone feel even slightly uncomfortable for any reason, chances are you probably won't be asked to do another project, or even another day in the studio.

It's hard work. That's not to say mixing a four- or five-hour gig isn't difficult, but there's the variation of mixing a whole set list of different songs plus the glory of the audience feedback. In the studio, the only feedback you get is from the producer and artist, and 99 percent of the time they're analyzing how you can make the mix better rather than singing your praises. And the level of concentration is definitely up a few notches. You can sometimes breeze through a gig, almost losing yourself in your mixing. In the studio, every moment of every track counts and requires your utmost attention.

Preparation. Live gigs may not even require a rehearsal to learn the songs. The studio mixer requires both system and personal preparation before even a single fader is raised, as you'll see in Chapter 3, "Mix Preparation."

Approach. Live mixers strive to get the same sound every gig, while a studio mixer strives to achieve a different sound on every song. Studio mixing requires experimentation and skill in working with constantly changing sounds and sonic characters, which is quite the opposite from a live mixer.

Pace. On a live gig there's a constant pace: show up, set up, soundcheck (maybe), gig, and tear down. In the studio the pace is usually set by a budget and/or a deadline. You may only have a limited amount of time to finish the mix, make any tweaks, and deliver it, regardless of whether the

mix feels finished or not. Or you may have enough leeway to work on a song for a few hours, put it away and work on something else for a while, then come back to it later.

The required skill set. For live mixing, the skill set requires that you know how to mix in an ever-changing acoustic environment and have a basic instrument/vocal balance technique. The studio requires your hearing to be more nuanced with a different reference point as to what sounds good or bad and how it will translate to other speakers outside the studio, plus you need a greater knowledge of what the gear and plugins are capable of.

The live wolf pack vs. the studio lone wolf. Most live performances require a group and a sizable supporting cast (unless you're working with a DJ or a solo singer/songwriter playing to tracks) that mostly stays the same every gig. Studio mixers are independent and usually work with different people on every project, or mix completely by themselves with little interaction with the client whatsoever.

Learning How To Mix

Mixing can't be taught; it has to be learned. Being a good mixer is the sum total of all of your mixing experiences. It's multiple "aha" moments that eventually get strung together. It's the sharing of experiences with your friends, clients, and books like this. It's the pat on the back and a "job well done" from the artist you tried so hard to please. It's the comparison of your mixes with ones you admire, only to realize that you're not as far away as you once were. In other words,

it's the experience of doing it over and over again, each time experimenting and refining until your ears sharpen and you develop a technique that you're personally comfortable with.

Becoming a good or even great mixer takes practice. There are a number of tips and tricks that you'll see in both this book and online courses like my Music Mixing Primer and Top 40 Mixing Secrets that can help accelerate your learning curve, but you still have to put in the time.

That said, a lot of different factors enter into a mix. The mix depends heavily upon the song, the musical genre, the musicians involved, the quality of the recording, loops or samples, and the arrangement—things that you may or may not have under your control.

There will be times when a well played, arranged, and recorded song will make the mix go so quickly and easily that you feel like a genius. Then there will be others when 100 tracks of poorly-conceived and shoddily-executed audio mush will bring you to your knees.

Both situations, as well as everything in between, are valuable. This is how mixing is learned, by experience.

If you're mixing mostly your own songs, make an effort to mix some of your friends' work as well, even as a trial, because every hour of hands-on experience will be a powerful learning tool.

Likewise, try to do mixes in different musical genres from what you normally work in. Different sounds and arrangements will open your ears to new sounds and techniques.

Above all, keep on mixing. It’s the best way to learn!

Monitoring

A mixer is only as good as the playback environment he or she works in, which is why this significant piece of of the mixing puzzle is examined before any specific mixing technique. If the monitors don't mesh with the environment or if the mixer doesn't interact well with either, then all the other tips and techniques may not help you as much as you'd like.

"I can use almost any studio as long as the monitors are good, because then at least you have the confidence in what you're hearing. It doesn't matter whether you're using crappy mics; if you're getting a sound that you like coming off the monitors, then you know it works."

—Ken Scott

The Listening Environment

Probably the single most important area that gets overlooked in most home studios is the listening environment. While it's possible that you can get lucky with a balanced, even sound by just setting up a couple of nearfield monitors in your room without thinking much about it, usually that's not the case. The average garage, living room, or bedroom was never intended as a listening space and has little in the way of acoustic treatment as a result.

Acoustically treating your room to smooth out some of the frequency imbalances can be cheaper and easier than you think, but that subject is beyond the scope of this book. You can read more on how to do that in my The Studio Builder's Handbook (Alfred Publishing). That said, here are a number of very simple things you can do to effectively improve your acoustic environment relatively cheaply and easily.

The Reflection Free Zone

The most critical part of creating a pleasing listening environment is creating what's known as a Reflection-Free Zone (or RFZ). This is an area around the listening position that tames the first reflections from the output of the speakers so they don't randomly bounce around the room (see Figure 2.1).

Diagram Description automatically generated

Figure 2.1 The Reflection-Free Zone

The concept is easy to implement. Sit in the listening position and have a helper move a mirror over the side wall. Everywhere that you can see the reflection of either speaker in the mirror requires acoustic treatment to tame the reflections. Repeat on the opposite wall and on the ceiling and you've defined your RFZ. It's a good idea to treat a larger area of the wall than you identify with the mirror so you'll be free to move around a little without leaving the Reflection Free Zone. Although it's more difficult to slide a mirror around on the ceiling, one way is to attach a hand mirror to a broomstick with rubber bands. Just by treating this area, you will improve your room by a surprising amount.

Acoustic Panels

Acoustic panels are the major way that reflections are kept from bouncing around the room when creating an RFZ. If your walls are hard (meaning there's no absorption), these reflections are going to cancel out certain frequencies from the direct sound of the monitors because of the standing waves, causing a number of unwelcome dips and peaks in the room's frequency response.

You can think of an acoustic panel as a very large picture frame that has sound absorbing material inside instead of a picture. Although you could permanently attach the sound absorbing material to the wall

(like most commercial studios do), using a sound panel allows you to move it as needed and even take it with you if you move to another house, apartment, or facility. Although you can build your own acoustic panels for very little money, they can now be purchased from a variety of sources online like Primacoustic, GIK Acoustics, ATS Acoustics and many more.

Overcoming Potential Acoustic Problems

Here are a few things to avoid if you can help it. These may seem like small fixes, but in some cases they can produce a dramatic improvement in what you hear.

Avoid placing speakers up against a wall. This usually results in some strong peaks in the low-frequency response. The farther away you can get from the wall, the less it influences the frequency response of your monitors, so the smoother that response can be.

Avoid the corners of the room. Worse than the wall is a corner, since it will reinforce the low end even more than when placed against a wall. Even worse than that is if only one speaker is in the corner, which will cause the response of your system to be lopsided.

Avoid being closer to one wall of the room than the other. If one speaker is closer to a side wall than the other, once again you'll get a totally different frequency response between the two because the sonic reflections from each wall is different. It's best to set up directly in the center of the room if possible.

Avoid different types of wall materials. If one side of the room contains a window and the other is painted drywall or something like carpet or acoustic foam, once again you’ll have an unbalanced stereo image because the absorption will cause one side to be brighter than the other. Try to make the walls on each side of the speakers the same in terms of absorption quality.

Monitors: Which Ones?

Which monitor speaker is best for you? There are certainly plenty of choices, and currently there’s no single favorite among the great mixers.

Probably as close to a standard as we’ve ever had was the Yamaha NS-10M, closely followed by the Auratone 5C (Figure 2.2). Because original NS-10s haven’t been made for quite some time now (except for the reissue clone CLA-10 from Avantone), their use has pretty much fallen by the wayside. Auratones have fallen out of favor since their peak of popularity during the ’70s, but many mixers still use them as an additional reference (although sometimes only one in mono in the center), especially now that they’re being manufactured again.

A picture containing electronics, loudspeaker, indoor Description automatically generated

Figure 2.2: An Auratone 5C

Courtesy of Auratone L.L.C.

That said, you don't necessarily need a set of monitors that are considered a standard or have even gained some popularity among mixers. It's possible to get great mixes out of virtually any set of speakers in just about any room, and that includes headphones as well. The trick is that you have to spend enough listening time to get a reference point as to what sounds good or bad when you play your mix back elsewhere. That's why mixers began to take their own speakers with them wherever they went (or asked for NS-10s) in the first place. It was a sound that they were familiar with, and since they were nearfields, the room didn't come too much into play during the mix, so they could be more comfortable with the result.

One of the good things about today's monitors is that since most of them now come with their own on-board amplifier, you no longer have to worry about buying an external amp to power them. That eliminates one of the major headaches of choosing a monitor, since the match with the amplifier is critical, and the speaker could sound very different when paired with a different amp. With the on-board amp now perfectly matched to the monitor, this is no longer a problem. Plus, the overall monitor package is less expensive as well.

That said, here are a few tips to consider before choosing a monitor.

Don't choose a monitor because someone else is using it. Monitors are a lot like guitars; just because Slash uses a Les Paul doesn't mean that it's also right for you. It might not match the music you make, or it might be too heavy for your particular body frame. Same with a monitor. Just because your favorite mixer uses a set of Dynaudio BM5As, that doesn't mean they'll be right for you, too. They may not fit the type of music you work on, they might be a bad fit for your room, or they may not even appeal to the way you hear.

Listen to the monitors before you buy them. Before a pro purchases a monitor, he or she will take some time to listen to them under a wide range of conditions. Why shouldn't you do the same? Yes, this may be a problem if you don't live near a big city with a pro audio dealer. Even if you do, you may not have a relationship with a dealer that allows you a personal demo in your own environment, but that shouldn't stop you from listening to them first. Many online dealers now have a generous return program where you can audition them for a time then return them for another pair if they don't suit your tastes.

This is a serious purchase, so don't take it lightly. Make the trip to your local pro audio or music store and prepare to spend some time listening. Listen to everything and spend as much time with each model as you can.

What should you listen for? Here's how to evaluate a monitor:

Listen for even frequency balance. While listening to a piece of music that you know well, check to see if any frequencies are exaggerated or attenuated. This is especially important in the midrange crossover area (usually about 1.5k to 2.5kHz). Try to focus on the cymbals on the high

end, vocals and guitars for the midrange, and bass mix element and kick drum on the low end.

Make sure the frequency balance stays the same at any level. The less the frequency response changes as the level changes (especially when playing softly), the better. In other words, the speaker should have roughly the same frequency balance when the level is quiet as when it's loud.

Make sure you have enough output level without distortion. Be sure that there's enough clean level for your needs. Many powered monitors have built-in limiters that stop the speaker or amplifier from distorting, but also may keep the system from getting as loud as you might need it to be.

Above all, don't buy a set of speakers without listening to them. It's usually very difficult for them to live up to your expectations if you've not heard them first. In fact, it's not a good idea to buy any monitor speaker unless you're really in love with it. You'll have to listen to these monitors for a lot of hours, so you might as well like what you hear.

Listen with source material that you know very well. The only way to judge a monitor is to listen to material that you're very familiar with and have heard in a lot of different environments. This gives you the necessary reference point to adequately judge what you're listening to. If you don't have anything that you've recorded yourself that you know inside and out, use a favorite CD that you consider to be well recorded.

TIP: Remember, don't use MP3s here! Use only CDs or a playback medium with an even higher quality 24-bit source, such as files on a flash drive or a personal digital recorder. That should give you a much better idea of the true frequency response of the system.

One thing I learned when writing speaker reviews for EQ Magazine over the course of five years is that you can easily get used to just about any speaker if you use it enough and learn its strengths and weaknesses. It also helps to have a solid sonic reference point that you're sure of to compare the sound with. For instance, if you know how things sound in your car, then adjust your mixes so they work when you play them there. Believe it or not, that's still a go-to place for many major mixers to reference their work.

"It still amazes me that with all these great and expensive tools, I can still throw it in the car and in like two seconds I immediately know what's wrong."

—Ed Seay

Basic Monitor Setup

Too often musicians and engineers haphazardly set up their monitors, and this is a leading cause for mix problems later on down the line. How the monitors are placed can make an enormous difference in the frequency balance and stereo field, and should be addressed before you get into any serious listening. Here are a few things to experiment with before you settle on the exact placement.

Check The Distance Between The Monitors

If the monitors are too close together, the stereo field will be smeared with no clear spatial definition, meaning that you'll not be able to pinpoint exactly where a mix element is panned. If the monitors are too far apart, the focal point or "sweet spot" will be too far behind you, and you'll hear the left or the right side individually but not both together. A rule of thumb is that the speakers should be as far apart as the distance they are from the listening position. That is, if you're 4 feet away from the monitors, then start by moving them 4 feet apart so that you make an equilateral triangle between you and the two monitors (Figure 2.3). A simple tape measure will work fine to get it close. You can adjust them either in or out from there.

Diagram Description automatically generated

Figure 2.3: Set the monitors in an equilateral triangle.

That being said, it's been found that 67 1⁄2 inches from tweeter to tweeter at the distance of a console meter bridge seems to be an optimum distance between speakers and focuses the speakers just behind your head (which is exactly what you want).

A real quick setup that we used to use in the days of consoles is to open your arms as wide as possible to each side and place the monitors at the tips of the fingers of each hand. This seemed to work well because of the built-in depth that the console meter bridge would provide, but it doesn't really apply in these days of workstations where the monitors are a lot closer to you. If that's your working situation, go back to the equilateral triangle outlined above.

Check The Angle Of The Monitors

Improper angling will also cause smearing of the stereo field, which could mean that you'll have a lack of mix element panning definition as a result. The correct angle is determined strictly by taste, with some mixers preferring the monitors to be angled directly at their mixing position and others preferring the focal point (the point where the sound from the tweeters converges) anywhere from a foot to about three feet behind them to eliminate some of the "hype" of the speakers.

TIP: A great trick for getting excellent left/right imaging is to mount a mirror over each tweeter and adjust speakers so that you can see your face clearly in both mirrors at the same time when you are in your mixing position.

Check How The Monitors Are Mounted

Monitors that are mounted directly on top of a desk or console meter bridge without any decoupling are subject to comb-filter effects, especially in the low end. That is, the sound from the monitor causes the desk or console to resonate, causing both the desk and speaker to interact as certain frequencies are either cancelled or reinforced. This is what's known as phase cancellation, and it causes a subtle yet very real blurring of the sound.

As a result, it will be just a little harder to hear your low end distinctly, which makes it more difficult to EQ. Phase cancellation can be more or less severe depending on whether the speakers are mounted directly on the desk or metal meter bridge, or mounted on a piece of carpet or similar material (which is very popular).

One of the quickest ways to improve the sound of your monitor system is to decouple your speakers from whatever they're sitting on. This can be done with a commercial product such as the Auralex MoPADs (see Figure 2.4), or you can make something similar relatively cheaply with some open-cell (closed-cell will work, too) neoprene or even some mouse pads. As mentioned earlier, carpet is sometimes used as well, although it's not nearly as effective as neoprene.

A picture containing electronics, loudspeaker, indoor, black Description automatically generated

Figure 2.4: Auralex MoPADs isolation pads

Courtesy Auralex Acoustics

It should be noted that recent studies have found that the differences between isolated versus un-isolated speakers aren't as great as once believed. It's still worth doing but it's probably a good idea to not splurge for an expensive solution when a cheaper one can provide more or less equal results.

Decoupling the subwoofers from the floor can sometimes prove to be worth the effort though. While sometimes the coupling with the floor can make your low end feel bigger, it will be a lot clearer and more distinct if decoupled. Auralex even has a product for this called the SubDude HD (see Figure 2.5), although you can probably put together a DIY setup that can work just as well.

Regardless of the brand, model, and type of speakers that you use, decoupling is a cheap and easy way to improve your sound right away.

A picture containing wall, indoor, electronics, loudspeaker Description automatically generated

Figure 2.5: Auralex Subdude HD with subwoofer mounted on top

Courtesy Auralex Acoustics

TIP: The best solution is to mount your monitors on stands just directly behind the desk or meter bridge. Not only will this improve the low-frequency decoupling, but it also can greatly decrease the unwanted reflections off the desk or console.

Check How The Monitor Parameters Are Set

Most powered monitors have anywhere from a single volume control to a wide array of parameter selections. Regardless of how many parameters your monitors might have, be sure that these controls are set correctly for the application and are set the same on each monitor (see Figure 2.6).

A picture containing text, electronics, black Description automatically generated

Figure 2.6: Monitor speaker parameter controls.

Courtesy Yamaha Corporation of America

Check The Position Of The Tweeters

Many monitors are meant to be used in an upright vertical position, but users frequently will lay them down on their sides. This results in a variety of acoustic anomalies that deteriorate the sound. If the monitors are laid on their sides, most mixers prefer that the tweeters of a two- or three-way speaker system be on the outside, thereby widening the stereo field. Occasionally, tweeters to the inside works, but usually it results in a smearing of the stereo image. Experiment with both, however, because you never know exactly what will work until you try it (see Figure 2.7).

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Figure 2.7: Tweeter position

Check The Desk Or Console Itself

If you're using a console, its angle; the type of materials used for the panels, knobs, and switches; the type of paint; and the size and composition of the armrest all make a difference in the sound due to reflections that cause phase cancellation. If the sound of the monitors mounted on top of the meter bridge is unacceptable, then try moving them toward you with extenders or put them on stands behind the console (don't forget to decouple them).

Three Steps To Adding A Subwoofer

It's not unusual for musicians and engineers who are doing a lot of work in their home studio on bookshelf-sized speakers to crave more bottom end. As a result, adding a subwoofer to their monitor system is the first improvement they think about. That's all well and good, but there are a few steps that you can follow that might help make your venture into low-frequency territory a lot easier.

A. Do you really need a subwoofer? Before you make that purchase, it's a good idea to be sure that a sub is actually necessary. Here are a couple of things to check out first:

Are you monitoring at a loud enough level? This is a trap that people with home studios fall into; they don't listen loudly enough, at least for a short period of time. First of all, if you listen too quietly, your ears begin to emphasize the mid-frequencies. This is great for balance but bad for judging the low end of a song. Crank up your monitors to a moderately loud level, at least when you're working on the low-frequency end of the spectrum. If you still don't have enough low end, go on to the next point.

Do you have an acoustic problem in your room? Chances are that either your monitors are too close to the wall or they're placed at a point of the room length where standing waves cause some of the low end to cancel out. This is more likely to be the cause of just one area of the low-frequency spectrum rather than the entire low end, though. Just to be safe, move your speakers a foot or so backward and forward and see whether you get some of the low end back. If not, move on to. . .

B. Purchase a subwoofer from the same manufacturer as your main monitors. The easiest way to get a smooth-sounding low end that doesn't cause you more grief than it's worth is to buy a sub to match the monitors that you use most of the time. That means if you're using JBL main monitors, choose a JBL subwoofer that's made specifically for that system. If you're using Genelecs, do the same, KRKs, the same, and so on. This will make a huge difference, especially at the crossover-frequency point where the mains cross over to the sub. It's usually extremely difficult to get that area to sound natural if you mix brands.

C. Calibrate your sub correctly. Most musicians and engineers who choose to use a sub just randomly dial in the level. You might get lucky and get it right, but it's more likely that your level will be off, causing a

number of unbalanced-sounding mixes until you finally figure it out. Here's how to calibrate the sub to your system:

Without the sub connected, send pink noise to your main monitors. At the listening position and while listening to one monitor only, use an SPL meter (just about any of them will do to get you in the ballpark, even a Smartphone app) and adjust the level of the monitor until it reads 85dB. The SPL meter should be set on C Weight and Slow. Repeat on the other channel and set that so it also reads 85 dB. Now your main monitors are calibrated level-wise.

Turn off the main monitors. Send pink noise just to the subwoofer. Set the level of the SPL meter so it reads 79 dB. Although it may seem like it will be lower in level, 79 dB works because there are fewer bands of low frequencies than the high frequencies (three for the low and eight for the high), so this SPL level takes that into account. You might have to tweak the level up or down a dB, but this will get you into the ballpark.

If there's a polarity switch on the sub, try both positions and see which one has the most bass or sounds the smoothest in the crossover area. That's the one to select.

If you follow these steps, you'll find that integrating a subwoofer into your system (if you decide you need one) will be as painless as possible.

Mixing On Headphones

Sometimes it’s just not possible to listen to your monitors to mix. When it’s late at night and your kids, significant other, or neighbor is in the next room separated only by paper-thin walls, you have no choice but to try to mix on headphones. Mixing on headphones does have its downsides, though:

You can’t wear them for as long as you need to (8, 10, 12 hours) before your head and ears get tired from the extra weight.

You have a tendency to turn them up, which can lead to some quick ear fatigue, again limiting your ability to mix for long periods.

Since most of the more expensive professional headphones really sound great, you get a false sense of what the mix is like (especially on the low end), and it causes you not to work as hard at getting the frequency balance of the mix right.

The vast majority of the audience won’t listen on real headphones after the mix is completed. Because a mixer is always aiming for a mix that sounds great on a wide variety of speakers on which the material is played, you want to stay in that realm if possible and even listen on some crappy speakers as a check if possible. Headphones (not to be confused with common earbuds) just sound too good for that.

That said, the latest room simulation software is a godsend to anyone who must mix on headphones for any reason. Plugins like Waves Abbey Road Studio 3 (see Figure 2.8), Sonarworks, and Acoustica

Sienna not only simulate some excellent sonic environments, but also have tailored EQ settings for most popular headphones.

A picture containing text, indoor, stereo, control panel Description automatically generated

Figure 2.8: Waves Abbey Road Studio 3 plugin

Courtesy of Waves

Even if you decide not to mix on headphones, they still do have their place in a mixing workflow. They're great for editing because they allow you to hear clicks, pops, and inconsistencies that you may otherwise miss while listening on speakers, and they're a great check for panning and reverb tails when mixing. That doesn't mean that you should use them for an entire mix, but if you have no choice, then by all means go for it. Just make sure that you listen to some other material that sounds great on your speakers first so that you have a reference point of what sounds good and what doesn't, and use one of the room simulation plugins if you can.

Headphone Mixing Tips

If you're going to mix on headphones, there are a number of considerations to be aware of regardless of the make and model that you use.

Headphone transducers won't reproduce the sounds of bass and kick drums in exactly the same way as the larger speakers in your room, which changes how you mix. With kick drum you will mainly be hearing attack rather than deep resonance of the kick drum, and with bass guitar you will mainly hear harmonics an octave above the fundamental rather than deep bass. However, room simulation plugins

like those mentioned above try to alleviate these frequency response problems.

Your ears will get quickly fatigued, so plan on taking breaks every 30 minutes or so.

As soon as you can, play your mix on another playback system at different levels to check where you're at.

These tips might not get you the best mix by themselves, but they can help you stay out of trouble so it takes less time to get the mix you want.

Speaker Calibration Technology

A number of companies now offer monitor speaker correction software that applies a detailed EQ curve to the output from your DAW in order to compensate for any deficiencies in your monitoring chain and environment. Some products are even intended to correct the frequency response of headphones as well as loudspeakers, as mentioned above.

The problem with using this technology with monitor speakers is that many of the problems that a mixer might encounter are a result of standing waves and other acoustic issues in the room. While this can be somewhat addressed for one position in the room, the problems reappear as you move about the listening area.

Headphones are a different issue as the acoustic issues (and the environment for that matter) are actually removed from the monitoring chain. While even an inexpensive pair of headphones can be equalized to become a rather neutral sounding monitor system, the question really becomes, “What is the correct EQ curve to apply?”

What the speaker calibration applications do is measure the complex interactions that take place between drivers, earcups, skull and ears. In order to get closer to the goal of headphones as a “reference monitor,” a profile of the exact headphones that you’re using is clearly beneficial, although most calibration packages allow for custom measurements to also take place and be stored in a user library.

Many mixers have raved about this technology, and it keeps on improving with each version. Chances are if it doesn’t reach to goal of improving your headphone environment now, it will in the future.

TIP: With any room correction or calibration software, the more samples you take in your room, the more likely you’ll get the results you’re looking for.

How Loud (Or Soft) Should I Listen?

One of the greatest misconceptions about music mixers (especially the great ones) is that they mix at high volume levels all the time. While some do, many mixers also find that they get better balances that better

translate to the real listening world by monitoring at conversation level (79dB SPL) or lower, although most will go up to a much higher level briefly when checking the low end of a mix.

Using high levels for long periods of time is generally not recommended for the following reasons:

First the obvious one: Exposure to high volume levels for long periods of time may cause long-term physical damage to your hearing.

High volume levels for long periods of time will cause the onset of not only ear fatigue, but general physical fatigue as well. This means that you might effectively only be able to work 6 hours instead of the normal 8 (or 10 or 12) that's possible by using lower levels.

The ear has different frequency-response curves at high volume levels that overcompensate on both the high and low frequencies. This means that your high-volume mix will generally sound pretty limp when played at quieter levels.

Balances tend to blur at higher levels, so what works at high volume won't necessarily sound great when played at a quieter level. However, balances that are made at quieter levels should always work when played louder.

Now this isn't to say that all mixing should be done at the same level and it should all be done quietly. In fact, music mixers (as opposed to

film, which always has a constant SPL level) tend to work at a variety of levels—up loud for a minute to check the low end, moderate while checking the EQ and effects—but the final balances are frequently done quietly.

"Generally speaking, when I put up the mix, I'll put it up at a fairly good level, maybe 105 [dB SPL] and set all my track levels and get it punchy and fun-sounding. And if I listen loud, it's only for very short periods of time. It's rare that I would ever play a track from beginning to end loud. I might listen to 20 seconds or 30 seconds of it here and there, but when I'm actually down to really detailing the balance, I'll monitor at a very modest level. I would say at a level that we could have a conversation and you could hear every word I said."

—Allen Sides

"I mix at different levels. I try not to mix too loud because it'll wear you down and fool your perspective. Sometimes it's very valuable to turn things down, but there's an up and down side to both. If you listen too soft, you'll add too much bass. If you listen too loud, you'll turn the lead vocals down too much."

—Ed Seay

"I'll monitor way loud to see what rocks. I'll monitor at a nominal level to get sounds together, then I'll monitor about 5 dB over background noise to bring all the elements into focus. If a mix works at 30 dB SPL, 25 dB SPL, it'll almost always work a lot louder. If you can hear everything at that low a level, then when you turn it up you'll have a very even balance. That's the way to get everything in the same plane, by listening extremely low."

—George Massenburg

"I listen quietly as much as I can. It's hard to check kick-drum level when it's quiet, so certainly you have to push it up every once in a while, but I fatigue pretty quickly when listening at loud levels. I can make better emotional and timbre decisions before I fatigue."

—Nathaniel Kunkel

Listening Techniques

Most experienced mixers have determined that regardless of how familiar they are with their monitors, they need some additional assurance that what they're hearing is really what they're hearing. As a result, a variety of listening techniques have been developed over the years that many still use.

Listening On Multiple Monitors

The number of monitor references that are used is an important aspect to getting the balance of a mix just right. Although a mixer may do most of his work on a single system, it's common to check the mix on at least one other speaker system as well. This might be the main soffit-mounted monitors if mixing in a commercial studio, the nearfield monitors of choice, and an alternative, which could be Auratones, NS-10s, computer speakers, or just about anything else. Couple that with a boombox, car stereo, or stereo in the lounge, and the average of all these systems should tell you what you need to know about the mix.

There are even some plugins that simulate all these different listening environments like Audified's MixChecker, which is worth checking out.

Most mixers will settle on a set of monitors they feel they can trust, learn its strengths and weaknesses, and then do a check on a smaller set. This can be anything from a small computer extension speaker, to earbuds, to even the speakers in a laptop.

The alternate speaker is used simply as a balance check to make sure that an element of the mix doesn't jump out as either too loud or too quiet in the mix. Also, one of the arts of mix balance is getting the kick drum and bass mix element to speak well on a small system, which is why an alternative monitor system is so important.

The second set of monitors doesn't have to be great. In fact, the worse they are, the better. Even a set of $10 computer speakers will do. The idea is to have another playback choice that will give you an idea of what things sound like in the every day listener's world, since unfortunately, there are a lot more people listening on lousy monitors than good ones these days.

In a studio, some mixers prefer to listen outside of the main listening area with the door open.

"What I'll do about an hour before printing the mix is prop open the control-room door and walk down the hall or into the lounge where the music has to wind its way out the door. It's real valuable to see if you hear

all the parts, and it’s real easy to be objective when you’re not staring at the speakers and looking at the meters.”

—Ed Seay

“I’m a big one for hallway. I like listening around the corner or on a blaster.”

—George Massenburg

“I’ll walk out of the control room and listen to it right outside the door. It’s interesting to hear what it sounds like through the crack in the door. Things pop out.”

—Joe Chiccarelli

TIP: Good television and commercial producers use a similar technique: Flip to a single mono Auratone (or similar small speaker), lower the volume to just perceptible, and see whether it still sounds like a record. Then raise the volume a tiny bit, walk out into the hall, and see whether you still like it.

Of course, the car still seems to be the gold standard of engineers, producers, and artists alike. Once upon a time it was important to hear what a mix it sounded like in that environment in the event that the song received radio play. Over time, the car stereo has evolved into a playback systemthat most people are very familiar with, considering the amount of time most of us still spend in our autos these days. With this in mind, don’t underestimate the power of using the car for checking a mix.

Listening In Mono

Sooner or later your mix will be played back in mono somewhere along the line, so it's best to check what it will sound like when that happens so you're not surprised later. Listening in mono is a time-tested operation that can actually give the mixer the ability to discern phase coherency, balances, and even panning.

The "classic" engineers who mixed back in the days of mono and made the transition to stereo all felt that the mono mixes they did were much more difficult than the stereo ones. In fact, back when stereo was still new, most of a mixer's time was spent on the mono mix and the stereo mix was done in just a few minutes afterwards. While it's true that there were far fewer mix elements to deal with in those days, the mono mix was still more difficult to achieve and took more time.

The point is that it's a good idea to make listening in mono an integral part of your mixing, since if it sounds good in listening with a single playback channel, it will most likely sound great with two!

Mono Selection

Mono playback selection is an integral feature on the mix buss of many modern DAWs, but not all of them. In some cases a plugin needs to be inserted into the mix buss to enable mono playback, but many mastering-type plugins also allow for mono selection. Luckily, there are

also some free plugins (like the Plugin Alliance bx_solo - see Figure 2.9) that do the job nicely.

Graphical user interface, application Description automatically generated

Figure 2.9: Plugin Alliance bx_solo

Phase Coherency

When a stereo mix is combined into mono, any elements that are out of phase will drop in level or even completely cancel out. This could be because the left and right outputs are wired out of phase (pin 2 and pin 3 of the XLR connector are reversed), which is the worst-case scenario, or perhaps because an out-of-phase effect causes the lead vocal or lead instrument to disappear. In any event, it's prudent to listen in mono once in a while just to make sure that a mono disaster isn't lurking in the wings.

Balances

Many engineers listen to their mix in mono strictly to balance elements together, since they feel that they hear the balance better this way. Listening in mono is also a great way to tell when one element is masking another.

"I listen in mono an awful lot and find it's great for balances. You can easily tell if something's fighting something else."

—Joe Chiccarelli

It's a good idea to switch to mono several times during the course of a mix just to make sure that every element can be heard clearly. Sometimes a quick check on a single channel playback will make a problem area much easier to pick out, or even identify one that you didn't know you had.

Panning In Mono (Yes, That's Right)

Although not many engineers are aware that their stereo panning can be improved while listening in mono, this is in fact a good way to achieve a level of precision not available in stereo.

"I check my panning in mono with one speaker, believe it or not. When you pan around in mono, all of a sudden you'll find that it's coming through now and you've found the space for it. If I want to find a place for the hi-hat, for instance, sometimes I'll go to mono and pan it around, and you'll find that all of a sudden it's really present, and that's the spot. When you go to stereo it makes things a lot better."

—Don Smith

3

Mix Preparation

Preparing for the mix can be as critical as the mix itself. A few simple procedures beforehand will make for a more comfortable and efficient mixing session that minimizes mistakes and hassles. This prep occurs before the first fader is raised, but sets the stage for a mix that flows better because you're not distracted by some rather mechanical steps that are better taken care of beforehand.

We can break the session setup into two distinct tasks: prepping your session and prepping yourself. Let's look at each.

Prepping Your Session

Prepping your mix has evolved over the years. At one time mix prep consisted of labeling the console, setting up the outboard gear, and biasing the tape machine. Today, mix prep is more about labeling the files and DAW channels, and arranging the session layout inside your DAW.

If we were to make a checklist of these steps, it would include:

o Make a session file copy

o Tweak the track timing

o Check the fades

o Eliminate noises

o Comp your tracks

o Tune your tracks

o Consolidate your tracks

o Arrange your tracks

o Make some pre-mix decisions

o Insert section markers

o Create groups and subgroups

o Create effects channels

o Assign channels to groups, subgroups and auxes

o Insert favorite EQs and compressors

That seems like a lot to do, but it's everything that's routinely done by mixers during the course of a mix, and it can more easily be completed before the actual mix begins. In fact, many A-list mixers use their assistants specifically for these tasks so they can concentrate on the more creative aspects of mixing. Let's take a closer look at each item on the list so you can see the details of each task is and learn why it's necessary.

Make A Session File Copy

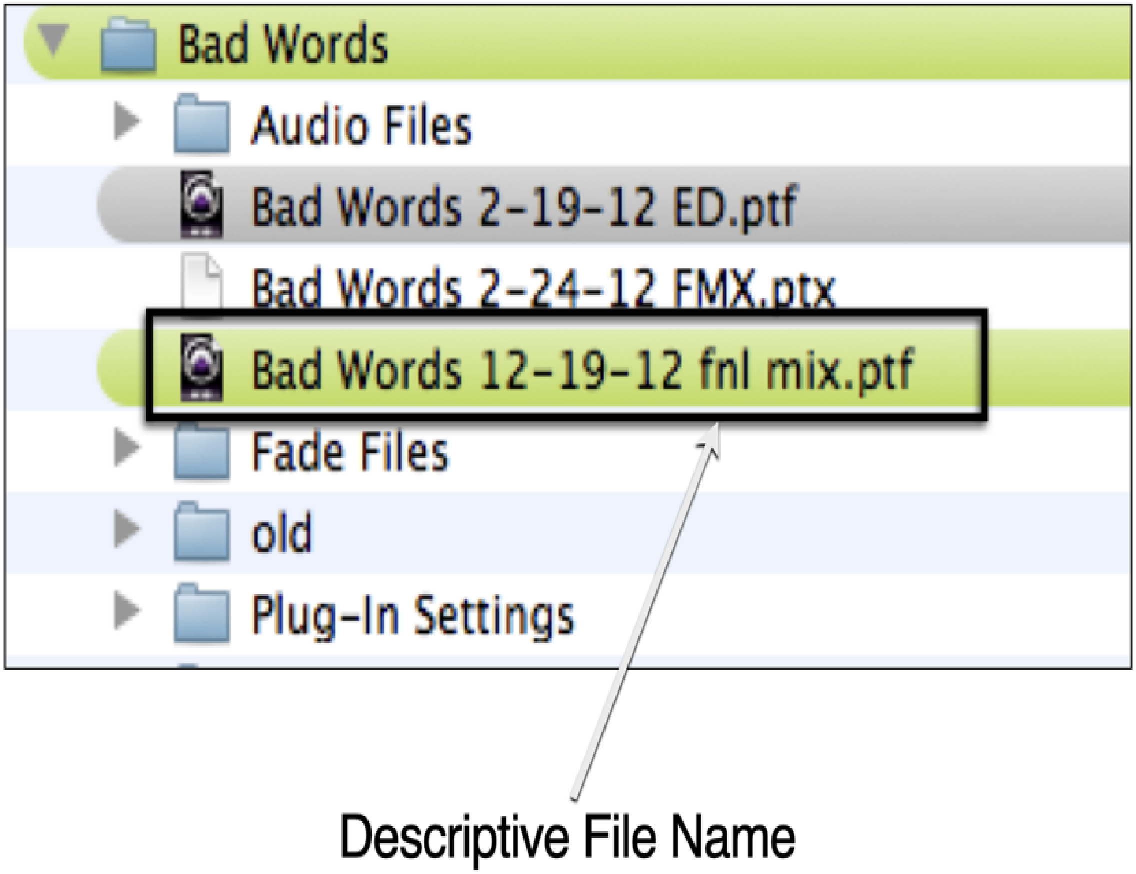


Figure 3.1: A descriptive file title

It's always a good idea to immediately make a copy of the original session file that you're given or have been using. That's because if you have to go back to where you started, if the file happens to become corrupted, or you just need to begin again with a clean slate, the original file will not be altered. When you make a copy of the session file, name it something descriptive like "songtitle mix" so it's easy to locate (see Figure 3.1). This keeps your previous session file safe if you ever have to go back to it.

Many mixers put a date in the file name, but that's not necessary as a time-stamp is almost always built into the metadata and can be easily determined by looking at the file info. It's not uncommon to have multiple versions of the same session during the same day; one way to differentiate one from another is to use letters of the alphabet or numbers at the end of the title, such as "Love Park mix 9-9-19a," "Love Park mix3," and so on.

If your DAW allows it, color-coding the file also makes it easier to identify. One way is to start with a preplanned series of colors that show the stage of completion such as blue for the beginning of a mix, red for a rejected mix, and green for when it's finished, or something similar to this approach. Of course, use whatever colors work for you.

TIP: It's also a good idea to make an additional copy of the session file on another hard drive, flash drive, online backup, or anyplace that you can easily grab it if for some reason your work file becomes corrupted.

Tweak The Track Timing

No matter how accomplished the players on a recording session are, many times there's some portion of a live track that doesn't feel quite right. While recording, however, you normally don't have enough time to have the musician play his or her part until it's perfect or punch in all the suspect parts as you go along.

Usually, the timing of the basic tracks will be tweaked right after your basic tracking session so you have a solid rhythm section to overdub against, but if you haven't done that or you're just now discovering some sections that don't feel right (which happens a lot), prepare for the joys of slipping and sliding time.

Of course, if most of the mix elements are samples or loops, then they're probably already aligned to the edit window grid, so all you have to worry about are the parts that have been played live.

You can read about some track timing techniques in the "Adjust the Timing" section of Chapter 12, "Advanced Techniques."

Check The Fades

Check each fade-in to be sure that a tail of a phrase isn't getting cut off too soon. Check each fade-out on elements such as background vocals or doubled instruments to be sure that the releases are the same. (You can read more about doing this in Chapter 12.)

Eliminate Noises

Now is the time to clean up each individual track. Although the noises might not be audible when played against all the other tracks, after everything is mixed and mastered you'd be surprised by how often something that was once buried can now come to the forefront and

bothers you. Also, by eliminating any extraneous noises, all the tracks magically sound more distinct and uncluttered. Remember, noise is cumulative. Follow these steps to eliminate an unwanted noise on a track.

Trim the heads and tails. Trim all the extra record time at the beginning and end of each track, regardless of whether it was recorded during basics or overdubs. Add a fade-in and fade-out to eliminate any edit noise (see Figure 3.2).

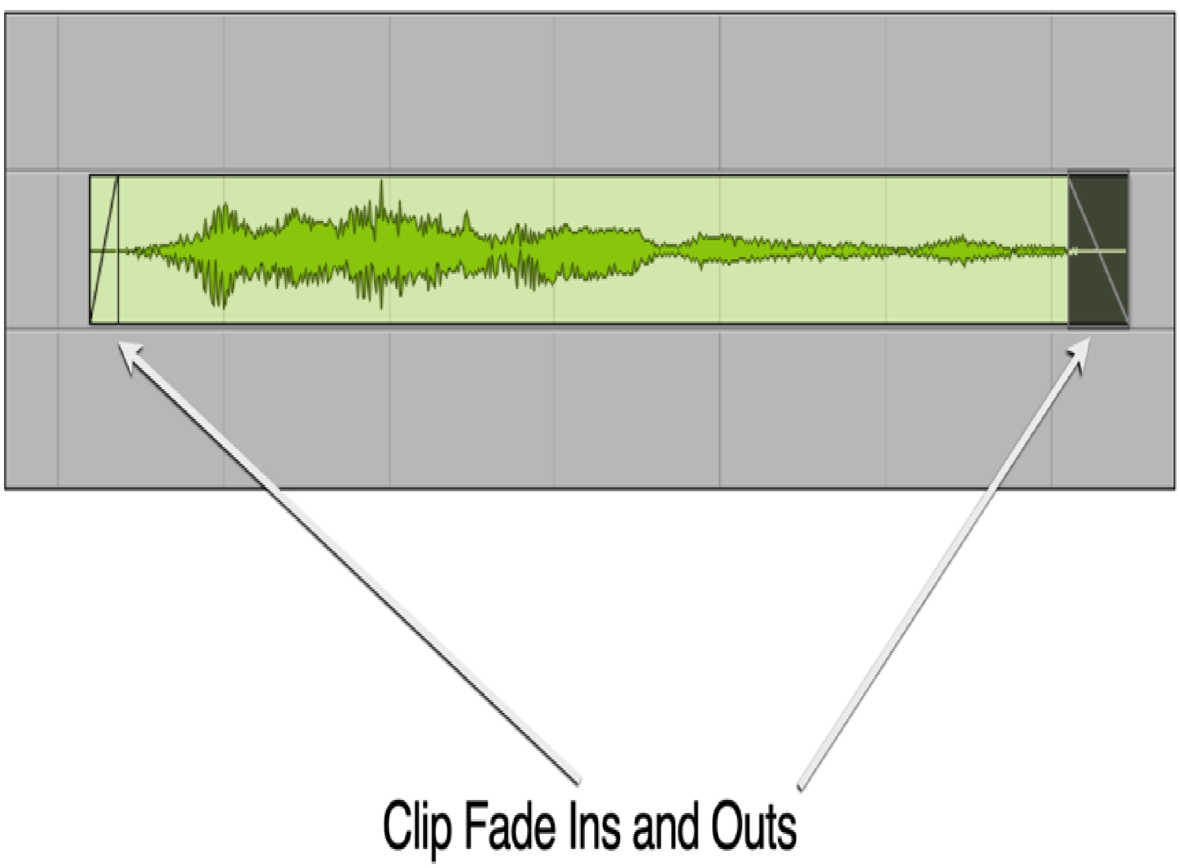
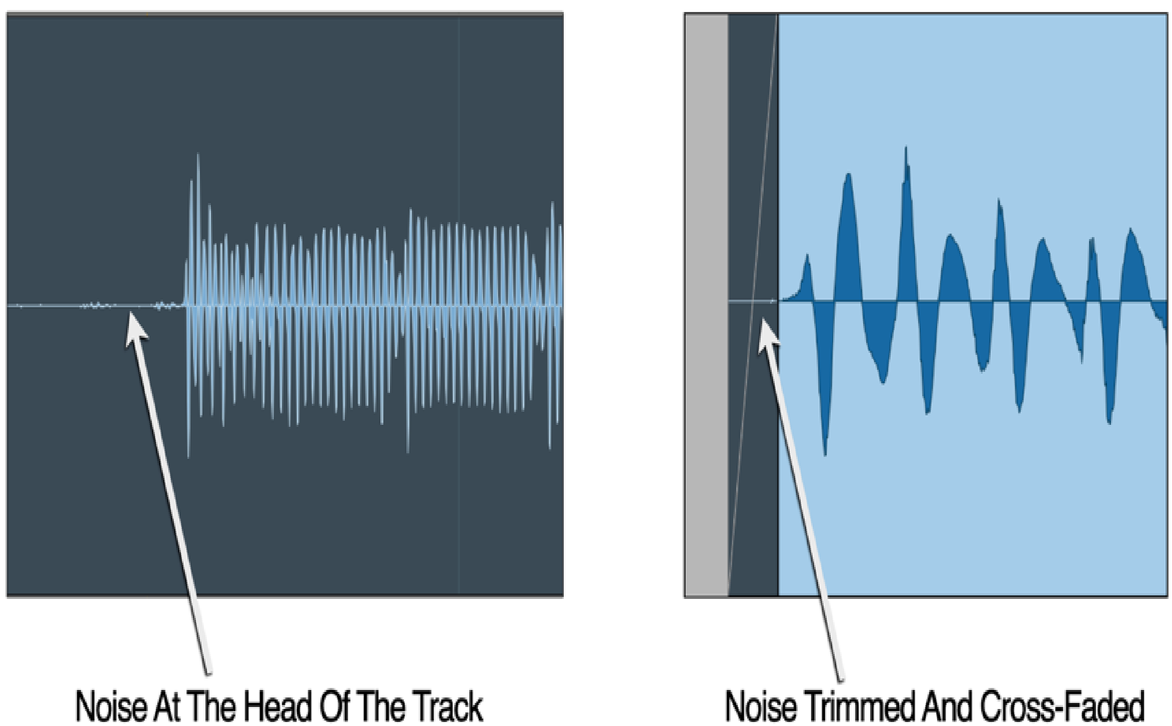


Figure 3.2: Fade-ins and fade-outs on a trimmed clip

Crossfade your edits. One of the biggest problems for A-list mixers when they get a session that's full of edits is that some of the edits click and pop because they don't contain any crossfades. Even if you can't hear a click or pop, it's a good practice to have a short crossfade on every edit to eliminate the possibility of an unwanted noise (see Figure 3.3). Again, these noises are cumulative!



A Trimmed And Cross-Faded Track Head

Figure 3.3: A trimmed and crossfaded track head

Delete extra notes from MIDI tracks. Delete any extra "split" notes that were mistakenly played. You might not hear them when all the instruments are playing, but just like the noise at the beginning of tracks, they have a tendency to come to the forefront after things get compressed.

"Today there's a lot of cleanup stuff that's sort of expected as part of the mixer's job, which should be a production thing. Production has gotten a lot lazier in the last 5 or 10 years. Now it's unbelievable how much garbage, extraneous stuff, clicks and pops, and unlabeled tracks that you get even in major projects."

—Bob Brockman

Comp Your Tracks

Comping shouldn't be left for mixing, as it's something that's normally taken care of directly after an overdub session where the vocal, guitar, or anything else is recorded using multiple takes. That being said, if you still have some vocal or overdub comping to do, now's the time. You can read more about comping techniques in Chapter 11, "Advanced Techniques."

Tune Your Tracks

Inevitably there’s always a note that’s a bit sour and needs tuning. Whether you use Auto-Tune, Elastic Audio, Melodyne or any other pitch correction plug-in, make sure that the timing isn’t thrown off when the note is shortened or lengthened. You can learn more about correcting pitch in Chapter 11.

Consolidate Your Tracks

Once all of your edits, comping and tuning is complete, it’s a good idea to consolidate all the edits of a track into a single solid clip. This means that instead of having a lot of edited regions on the track, you have one solid track with no edits (this can usually be undone, by the way). Not only does this make the track easier on the eyes, but it will keep you from accidentally moving or deleting an edited clip during the mix. Be aware that this might not be an advantage if you’re planning on changing the level, effect or plugin just on a single clip later during the mix — assuming your DAW has these features. That said, most DAWs allow you to backtrack or un-consolidate if necessary.

Arrange Your Tracks

These days, a typical session has plenty of tracks that won’t be used in the final mix. Deleting or hiding these tracks and then assembling the rest in a logical order can be the single most useful thing you can do while prepping your mix. Here are a number of steps to take if you’ll be

mixing the song yourself, or even more important, if you'll be giving it to someone else to mix.

1. Delete Empty Tracks

Empty tracks take up space in your edit and mix windows without adding anything useful, so it's best to delete them. During tracking or overdubs, it frequently makes sense to have empty tracks readily available so you can go instantly to another take or overdub with a minimum of time However, if you've gotten to the mix stage without using some of them, they're only taking up valuable space on your desktop, making it more difficult to see some of the channels you really need. Delete them.

2. Deactivate And Hide Unused Tracks

Any tracks that you know won't be needed (sometimes they're marked or color-coded beforehand) just soak up your computer's system resources. Even with a power machine, these resources may become a precious commodity, especially if you end up using a lot of DSP-intensive plugins during the mix. Deactivate the tracks and then hide them from the timeline and mix panels so they don't distract you and take up desktop space.

3. Reorder Your Tracks

Reordering tracks into logical groups of instruments or vocals makes finding individual instruments or vocals easier during the mix. The idea is to group any similar instruments or vocals together, so all the guitars are next to each other, the drums and percussion are next to one another, and keyboards, horns, strings, and all the vocals are together. This makes it easier not only to find a track, but also to group them later if needed.

4. Color-Code The Tracks

This isn't absolutely necessary, but it does make things easier to find if your DAW app has this ability. For instance, all the drums might be red, guitars blue, the vocals yellow, and so on.

5. Correctly Label The Tracks

Many workstation apps automatically assign a name to any new track that has been recorded, but usually the names don't relate to the mix element (see Figure 3.4). It's really easy to mistake one track for another and turn a fader or parameter knob up and up and wonder why nothing is happening, only to find that you're tweaking the wrong track. That's why it's important to clearly label each track that has a name like "gt166" to something descriptive like "guitar" or " rhythm gtr." Consider this to be the same as identifying the channels of a console with console tape, and make it a habit when prepping.

Relabeled Tracks

Figure 3.4: Relabeled tracks

Even if it's your own project - don't assume that you'll be the only one to ever mix it. Label each track so that anyone can easily identify it. Labels such as "John 1" and "Paul 4" don't mean anything to anyone but you. Who's John? What is Paul playing or singing? Mark each track logically.

"One issue is organization, because the track labeling is often really poor, and I find I'm spending hours of prep time before I can even get into a mix. I get lots of tracks that are just labeled 'Audio 1,' 'Audio 2,' or 'Bob 1' or 'Bob 2.' I don't care that it's Bob singing. I need to know exactly if it's a high harmony in the chorus."

—Joe Chiccarelli

Make Your Decisions

One of the unfortunate aspects of our digital world with its unlimited tracks is that it's really easy to put off decisions about which tracks to use until the mix. This can be difficult even if you're familiar with all the tracks, but nearly impossible if you're seeing the session for the first time.

For instance, if you have seven different guitar takes, you don't need to keep them all. Make a decision about what will work best and then deactivate and hide the others. If you have four mic setups for each guitar, trust me, you don't need them all. Make a decision about which one will work and save yourself a lot of time. Chances are that the difference between them isn't as great as you thought it would be.

"Make decisions as much as you can along the way. Everyone is too willing to let any decision wait until the last minute instead of making it on the spot. Today you see people piling on track after track and waiting until the mix to sort it out. That makes everything take a lot longer and makes it harder to mix because you're not sure how everything is going to fit together."

—Ken Scott

"That's the thing that does take a bit more time these days in that there are so many possibilities and people don't make decisions now because they know they don't have to commit to something until the very end."

—Joe Chiccarelli

Insert Section Markers

Time markers (sometimes called memory locations) are a major timesaver in any DAW and essential for an efficient mix. If you haven't done this already, now is the time to mark each section of the song (see Figure 3.5). Most veteran mixers insert a marker a bar or two before each new section so there's some pre-roll. Also make sure that other points in the song, such as drum fills, accents, or even the halfway point in a section, are clearly marked as well.

Figure 3.5: Marker or memory locations

Create Groups And Subgroups

Subgroups are a way to pre-mix a number of channels before they're sent out to the main outputs of the mixer (see Figure 3.6). This allows you to not only control the level of several channels with a single fader, but also add EQ or effects to all of those channels via the subgroup channel as well.

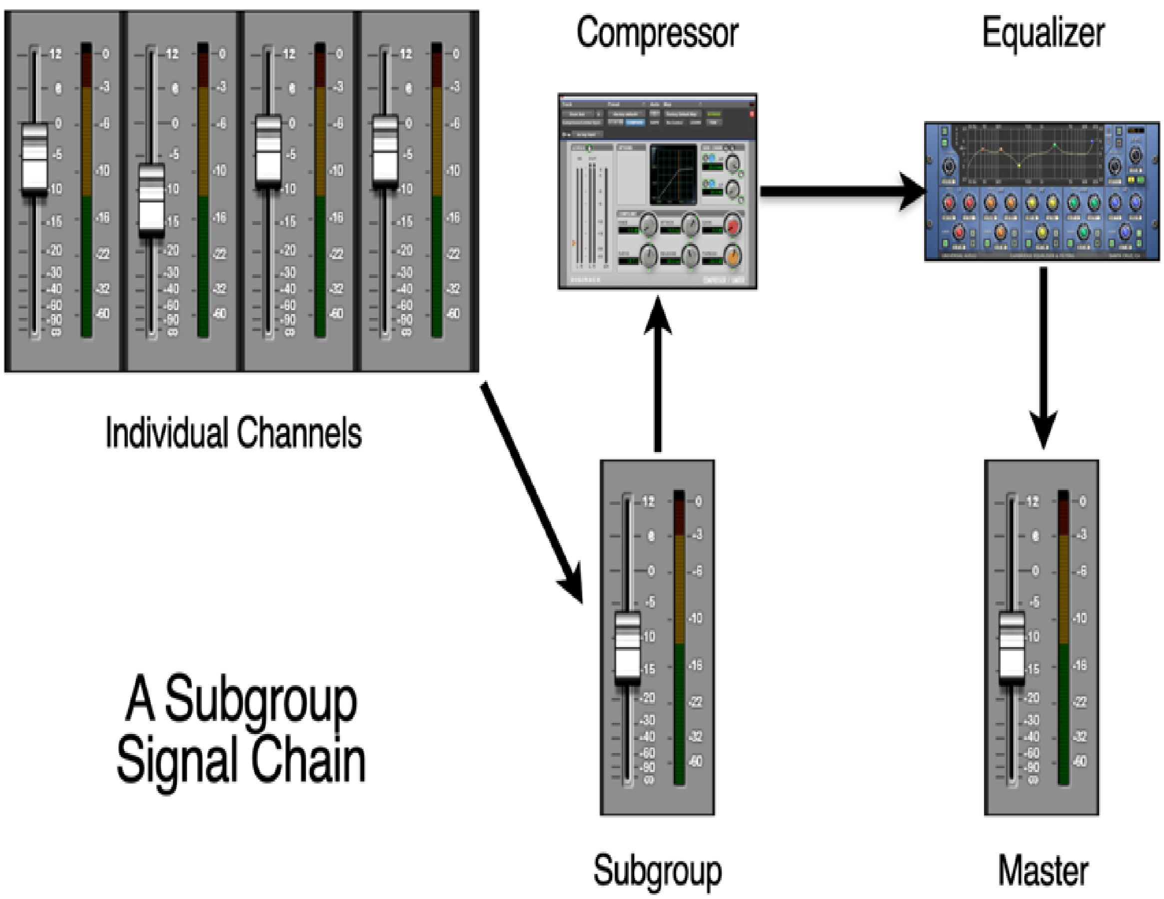


Figure 3.6: Subgroup diagram

Subgroups can be useful during mixing because they allow you to group similar elements of the mix so you can make adjustment by instrument sections, rather than individually (see Figure 3.7). A mix can go much faster if the subgroups are created and the particular instrument or vocal channels are assigned to them ahead of time.

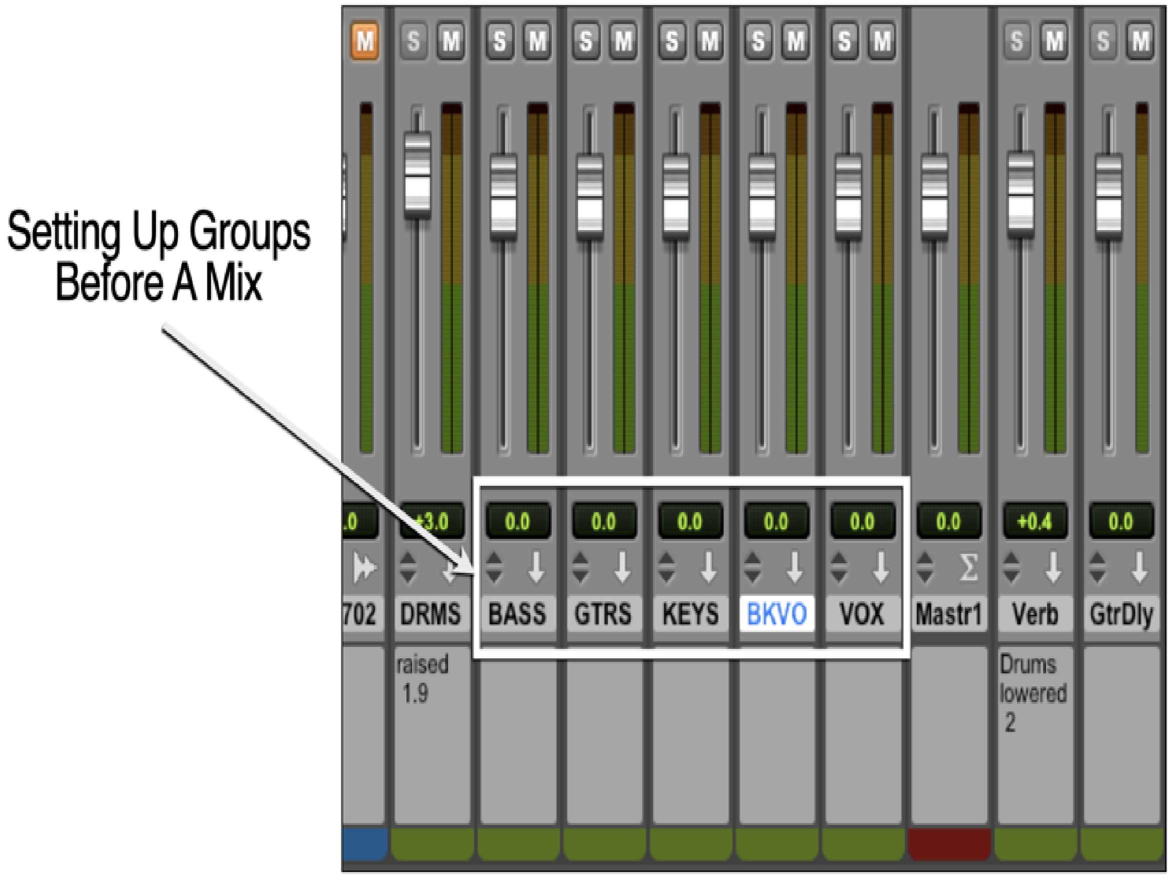


Figure 3.7: Setting up subgroups

Typical groups might include drums, guitars (if there's more than one or there's two tracks for stereo), lead vocals (if there's a double or there's a track for each song section), background vocals, horns, strings and synths, or any instrument or vocal recorded in stereo across two mono channels.

Groups (not to be confused with subgroups) are virtual subgroups in that the assigned faders are linked together so that when one is moved, they all move in proportion to their balance, but they still retain their individual outputs to the stereo master buss (see Figure 3.8).

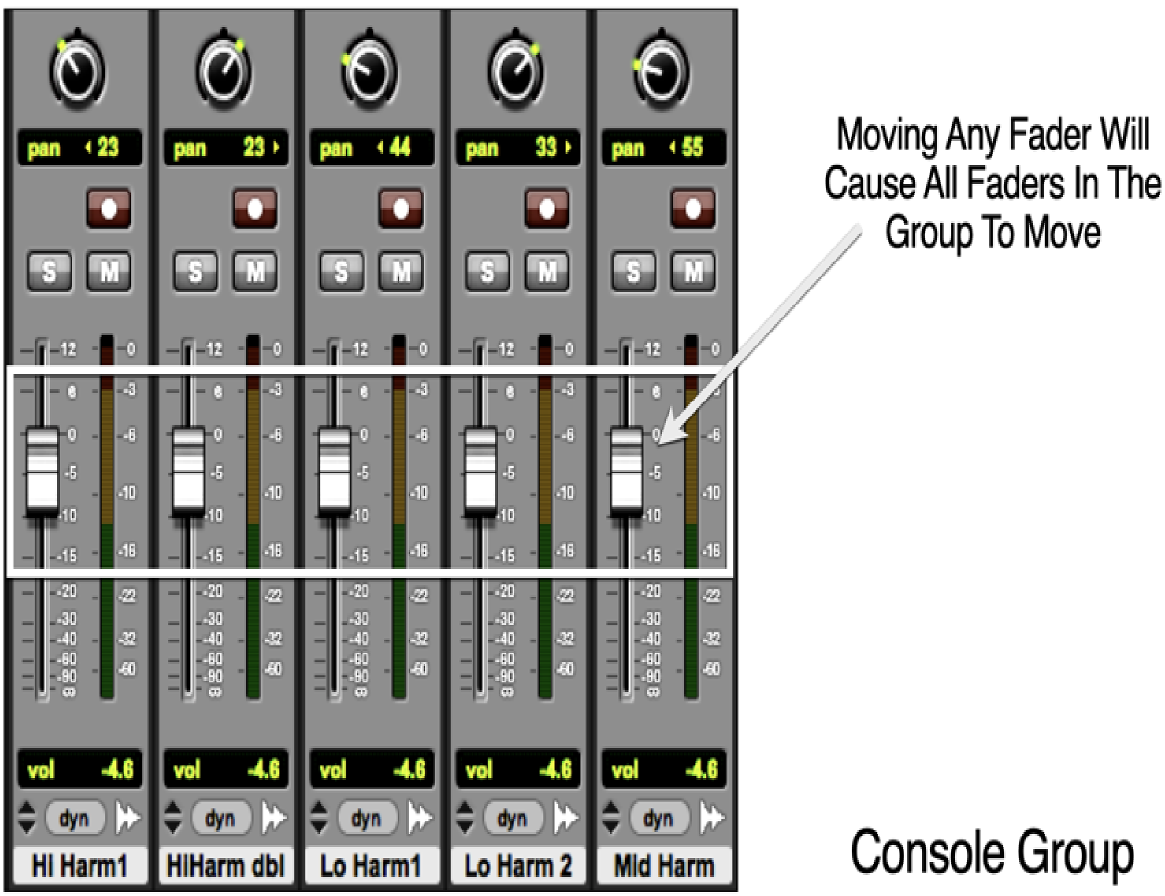


Figure 3.8: Group diagram

Generally speaking, groups work well with mix elements that were recorded as stereo onto two separate tracks, such as piano, organ Leslies, guitars recorded with multiple mics, drum overheads, and the like. Subgroups work best with large groups of instruments such as the drum kit, background vocals, and string sections, although how they're used is purely up to the taste of the mixer. Groups may be used within subgroup channels, as illustrated in Figure 3.9.

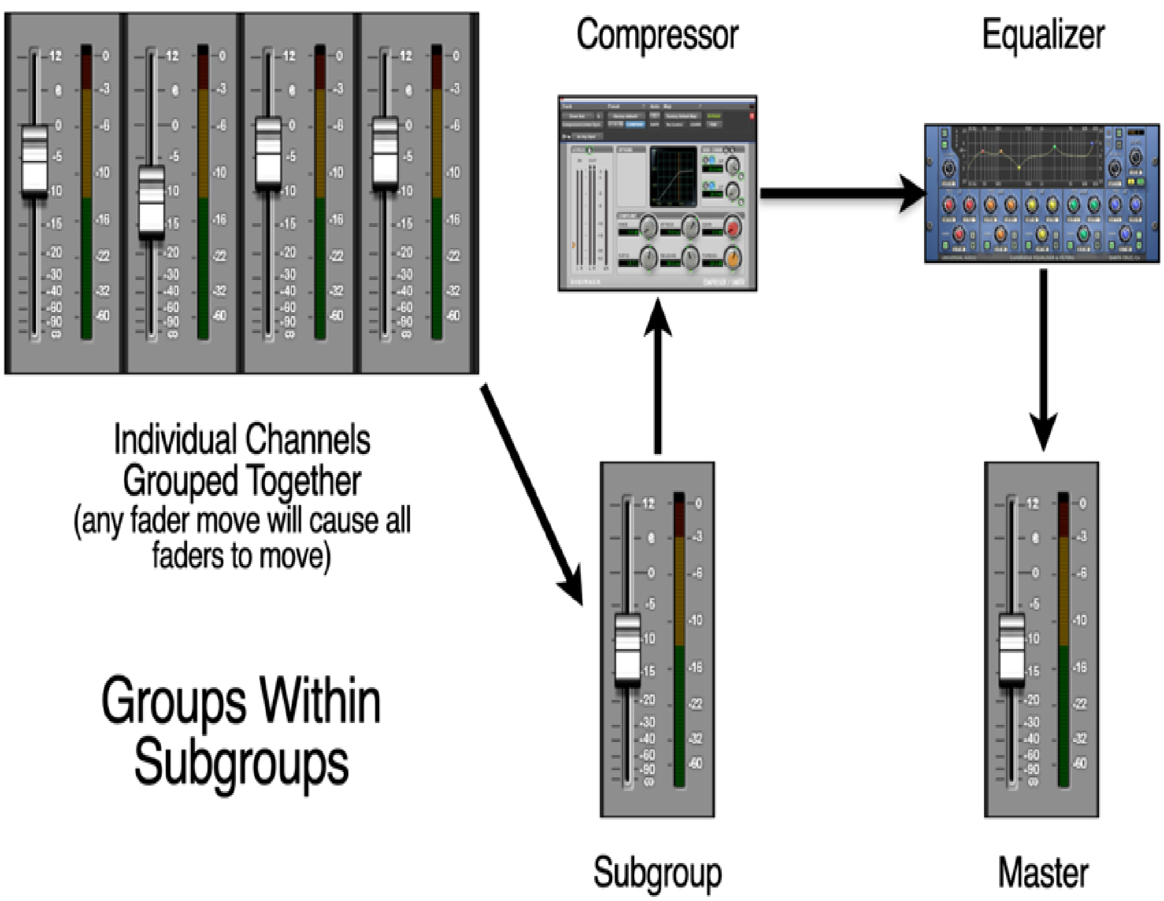


Figure 3.9: Groups used within subgroup channels

Create Effects Channels

Many mixers have a standard set of go-to effects that they'll set up before they begin a mix, since they know that they'll eventually be put to use at some time during the mix. This will be covered more in Chapter 8, "The Dimension Element," but one quick setup that works well even for tracking, overdubs, and rough mixes utilizes two reverbs and a delay and is set as follows:

For drums: Use a reverb set to a dark room sound with about 1.5 seconds of decay and a pre-delay of 20 milliseconds.

For all other mix elements: Use a plate with about 1.8 seconds of decay and a pre-delay of 120 milliseconds.For vocals: Use a delay of about 220 milliseconds with two repeats.

These settings work very well together and create a nice blend of ambience without much tweaking. Another common setup uses two reverbs and two delays, which are set like this:

Short reverb: A room program with the decay set from .5 to 1.5 seconds of decay with a short pre-delay timed to the track. (See Chapter 8 for more detail on how to do that.)

Long reverb: A plate or hall program with a decay set from 1.5 to 4 seconds of decay and a pre-delay of as little as 0 or as much as 150 milliseconds timed to the track (depends on your taste and what's right for the song).

Short delay: A delay of about 50 to 200 milliseconds.

Long delay: A delay from about 200 to 400 milliseconds.

Your own particular starting point might use a lot more effects, or you may prefer to add effects as they're needed during the mix. Regardless, it's a good idea to have at least some effects set up before you begin the mix so you won't break your concentration on the mix to set them up later. Some additional effects setups are described later in Chapter 8.

Assign The Channels

Usually, you know ahead of time that certain tracks will require a particular effect (such as a short reverb decay on the drums or snare). It's best to assign those channels to the appropriate sends and pan them accordingly before your mix begins, but remember to make sure that the send is set to infinity or off before you start.

TIP: Randomly assigning busses to groups or sends can get confusing on a large mix. Some mixers choose to group the busses by function. For

instance, Busses 1 to 10 for subgroups, 11 to 20 for reverbs, 21 to 30 for delays, and so on.

Insert Compressors And Limiters

In most modern mixes at least a few channels, such as the kick, snare, bass, and vocal, will usually need a compressor during the mix in order to modify the track's dynamic range. The mixing process can be sped up if the compressors are inserted on those channels ahead of time during mix prep on DAWs where a channel strip isn't already pre-inserted. Remember to leave any compressor or limiter bypassed until you decide to use it later during the mix (see Figure 3.10).

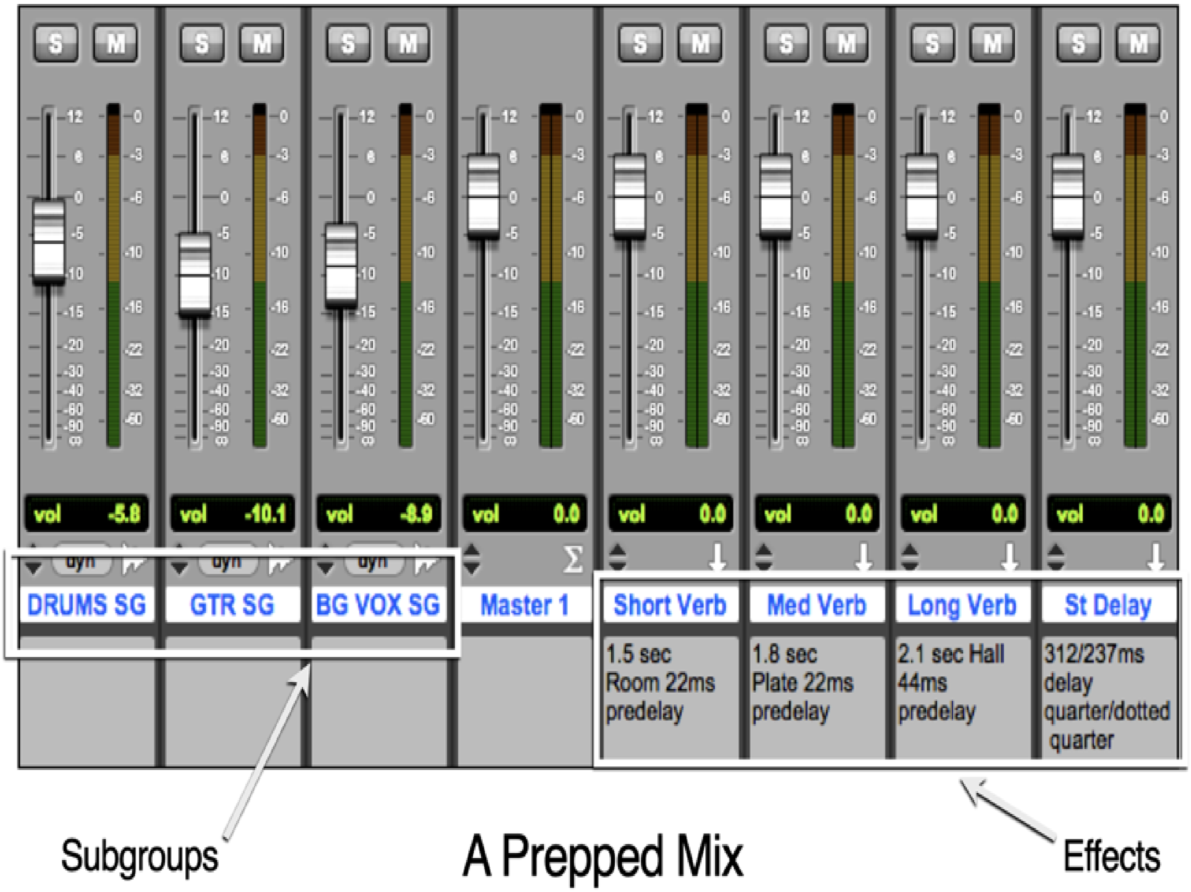


Figure 3.10: A fully prepped mix

Personal Preparation

One overlooked area of mix prep is personal preparation, which means getting yourself into the best physical and mental state to mix. Each mix requires focus and concentration, and this is where you get yourself into that proper headspace.

Calibrate Your Hearing

One of the biggest attributes for an engineer or producer to develop is what's known as "studio ears." That's the ability to discern between minute changes within the music that you're hearing. Is that guitar slightly out of tune? What frequency is the vocal harshness coming from? Is the keyboard part coming in slightly early or late? The better your studio ears, the more finely you can hear these irregularities.

Our ears are actually amazing organs that are capable of hearing sounds so faint that they move the eardrum less than the diameter of a hydrogen molecule. It's important that we first calibrate them to take advantage of their enormous capabilities. Here's how:

Try to stay in the quietest area that you can for as long as possible before you begin your mix.

Concentrate on the sounds that you're hearing and try to identify what they are and the direction they're coming from. Studies have found that this can make your hearing much more acute. What are the five loudest sounds you can hear? What are the five secondary sounds that you can hear? Are there additional sounds or noises that show up occasionally in your everyday life?

Stay away from a large meal before you mix. Studies have shown that your hearing may be temporarily less sharp during digestion.

To improve your ability to hear faint sounds, relax your jaw or just smile. There are tiny muscles in your jaw that can actually disrupt the action of your eardrums and Eustachian tubes, which control the inner-ear pressure. Try it now. Smile! Can you feel the effect this has on your ears?

If you will be doing work that requires your attention on a computer monitor, even small noises can subtly blur (distort) your vision. Turn the monitor level down and try to keep the uninvited noises at bay.

TIP: Also remember that closing your eyes while mixing can sometimes improve your hearing by both lessening the distractions and allowing your brain to concentrate additional processing power on that sense.

"I've been mixing with my eyes closed for most of my career. I think I have an easier time visualizing the three- dimensional panorama that's coming out of the speakers. Somehow when I close my eyes it's easier for me to see an instrument or vocal by removing my eyes from the equation altogether."

—Bob Brockman

Find Your Listening Reference Point

Even if you know the sound of your studio very well, it's still important to establish your listening reference point before you begin. This is accomplished by playing at least one recording or mix that you know well so you have a reference point as to what the room and monitors sound like. Listening to a mix or two will also calibrate your ears to the listening environment, which will help to keep you from over- or under-EQing as you go along.

Prepare For Note-Taking

During the course of a long mix, you'll probably have to take some notes, so have a pen and a pad of paper or even some Post-it notes ready to write on. If you're using a hardware controller, you'll need a roll of console tape. Shurtape FP 726 is the industry standard that can be reapplied to mark the names of the channels without leaving any sticky residue behind.

Make Yourself Comfortable

Most mixes take a while, so both you and your environment need to be comfortable. Make sure your clothes and shoes are comfy, the room temperature is set just right, and the lighting is adjusted so you can easily see any monitor screens that you may be using without squinting or glare, because this can lead to eye strain and overall fatigue before you realize it. It's also a good idea to have some beverages and a snack ready for later when you need a break.

TIP: A comfortable chair can provide a greater feeling of well-being, which in turn can heighten your senses, like your most important one for mixing: hearing.

Take Frequent Breaks

Be sure to take frequent breaks while you're mixing. This will not only give your ears a chance to rest, but also prolong your ability to work. Some mixers may take a break as frequently as every hour, while others might wait for three or four hours before they feel the need to rest.

A break means a complete break from the listening process, not just from mixing. Leave the room. Get some fresh air or sunshine, grab a beverage, and change the subject for 10 or 15 minutes. This will go a long way to a successful mix without you feeling worn out at the end.

Stay Focused On The Mix

The studio isn't a place for playback parties with your friends or potential clients. Every time you stop for as little as a phone call, it breaks your concentration and it becomes that much harder to get the momentum back. Turn the phones off and check your messages only during breaks. Even then, try to limit return calls to emergencies. Anything that breaks your concentration takes your focus away from the mix.

It's especially difficult to stop a mix to play back a track for your friends, who of course will then want to play you one that they're working on. Not only is your focus completely shot, but your ears need to be recalibrated as well. Stay disciplined. Tell your friends that you'll be happy to meet them for a playback session after your mix is completed.

Setting up for a mix is a lot more work than you might have thought, but it's time put to good use. Once these preparations are out of the way, your files, tracks, mind, and ears are all set for the mix ahead.

TIP: Remember the mantra - "Fix it before you mix it!"

4

The Mechanics Of Mixing

Although most engineers ultimately rely on their intuition when doing a mix, there are certain mixing procedures that they all consciously or unconsciously follow. That said, there are a couple of basic physical concepts of mixing that are essential for a big sounding yet clean mix as well.

Level Metering

Until recently mixing was fairly simple from a metering standpoint — keep the VU meters out of the red and everything should sound fine. Today we're faced with a wide range of metering options thanks to digital audio workstation mixing and this array of options can make something that used to be fairly simple much more complex, at least at first. Let's look at the various types of level meters, when they're used, and why.

Some Metering History

The standard VU (Volume Unit) meter was designed in 1936 in a joint effort by CBS, NBC and Bell Labs and was originally called the SVI, or Standard Volume Indicator. Since the meter scale was calibrated in Volume Units, that's the name that stuck and became the analog VU meter that we're used to seeing on all sorts of professional analog audio gear to this day.

The VU meter was fairly cheap to implement, but it had limitations. For one thing it was rather slow to respond to a signal. That meant that a mix element with fast transients (like drums or percussion) could provide peaks that were as much as 10dB or higher over what the meter was indicating. Also, the meter was built to provide an RMS (an electronic calculation called Root Mean Square) which is an average value of the signal voltage. The reference level was 0VU (zero Volume Units), and anything above this signal level was in the red zone and indicated a potential overload. This may have been true for some inexpensive prosumer type gear with low headroom but not necessarily for professional gear with a great deal of headroom. Later versions of the VU meter also included a peak LED that lit when a predetermined transient level was detected.

Because of these limitations there was a desire by many broadcast mixing engineers for a meter with a faster response. That meter was the PPM (Peak Program Meter), which actually began development before the VU meter in 1932. PPM meters didn’t actually measure the signal peak but they were a lot better at detecting those fast transients than normal VU meters missed. They were also much more expensive than VU meters, which is one of the reasons they never found widespread use in recording studios.

As solid state electronics became more mature and LEDs became cheaper and easier to use, LED segment meters gradually replaced VU meters on all types of pro audio gear. Instead of VU-style ballistics, these new meters acted more like peak meters in that they indicated a signal level up to 0 headroom (known as 0dBFS for Full Scale) with a red overload indicator, which is what we mostly use today in the digital domain.

Types Of Digital Level Meters

While the top of a meter scale is usually 0dBFS with an overload indicator above it, the rest of the meter can be calibrated in many different ways to help us view the signal more appropriately depending upon the need and the situation.

Signal Present Meters

On audio hardware where traditional analog meters are too expensive to include or where there's not enough room on the faceplate, signal present meters are included instead. This can be anywhere from one to three LEDs that indicate when a signal is present (green), the signal has exceeded recommended levels (orange or yellow), and overload or clipping has occurred (red). Sometimes a single multicolored or tri-colored LED is employed. Also, many plugins feature a variation on this theme where more sophisticated metering is not considered essential.

Peak

The peak value is the highest level that the waveform will ever reach, like the peak is the highest point on a mountain. A peak meter is great protection in preventing overloads since the meter will provide a true indication of exactly where the signal peaks are. Peak meters are designed to respond quickly to the signal level, although peak meters from manufacturer to manufacturer may have different ballistics.

RMS

RMS meters are designed to approximate the way your ear perceives sound levels. Since your ear will typically not perceive sharp peaks to be as loud as they really are, an RMS meter measures the average level of the signal. This type of meter provides a rather slow measurement as it averages out the peaks and troughs of the waveform. Although the RMS meter attempts to measure loudness level of a program, it’s not nearly as good at doing so as the more up-to-date LUFS meter (below).

K-Scale

K-Scale metering was developed by mastering engineer Bob Katz as a metering and monitoring system designed to discourage extreme program loudness while providing consistent monitoring levels. It combines the characteristics of both a peak and an RMS meter in such a way that the control room monitors can be calibrated so that when the level meter reads “0,” the sound pressure level in the room would be 85dB SPL.

There are three K-Metering scales: K-12, K-14, and K-20. Each scale provides a different amount of headroom. For instance, the K-20 scale provides 20dB of headroom while the K-12 provides 12dB. The K-System’s meters have three, color-coded regions - green, yellow and red. Green corresponds to quiet, yellow to loud and red to very loud.

The reason for the different scales is that different types of music require a different calibration. Pop recordings might not need much dynamic range, so a K-12 scale would be used, while a K-20 scale might be used for classical or jazz where more dynamic range is preferable.

While an excellent idea, K-scale metering never caught on with most music mixers, although it's provided as a metering alternative in many modern DAWs.

LUFS/LKFS

Over the last few years, you might have noticed the differences in levels between television shows, commercials, and channels have become pretty even, with no big jumps in volume between them. That's because viewers were complaining for years about the dramatic increase in level whenever a commercial aired because it was so compressed compared to the program that you were watching. Congress set out to do something about this, and in 2012 adopted a method to normalize those volume jumps similar to what the European Broadcast Union put into place the year before - Loudness Unit Full Scale or LUFS.

LUFS (called LKFS in Europe) is a way to measure the perceived loudness of a program by measuring both the transient peaks and the steady-state program level over time using a specially created algorithm. It's different from a normal DAW peak meter in that it doesn't represent signal level – it measures how loud we perceive an audio program to be. For a broadcaster this is actually pretty serious, since if a station violates the mandated LUFS level of -23, it could possibly lose its broadcast license.

Even though LUFS was intended primarily for broadcast audio delivery, it has a new increased meaning in music production as well. Thanks to the fact that streaming services like Spotify and Apple Music are now

normalizing songs so that the level is the same from tune to tune, there's no real benefit for compressing a song to within an inch of its life any more. In fact, less volume and more dynamic range are actually your friend.

Using a LUFS meter allows you to optimize your music mixes for a variety of platforms to be sure that they're always in the sweet spot for dynamic range. We'll discuss the use of the LUFS meter more in Chapter 12.

Other Types Of Level Meters

Believe it or not there are other types of metering selections also available, including five versions of digital PPM metering, Linear, Linear Extended, and more. All are designed to do the same thing - tell us the level of the signal and measure how close to an overload it is. The way this is displayed may be different, but they all serve the same purpose.

Which One To Use?

Now you may be asking yourself, "So with all these metering choices, which one is the best to use?" For some, meter choice is a function of having used a certain type of legacy meter in the past, feeling comfortable with it, and wanting to continue to use it in a digital workstation. For others it's a function of the type of mixing environment and application, such as broadcast or postproduction. Usually the default metering for most DAWs is peak and that's probably the best to use unless there's a specific reason to change. It's also the safest from the standpoint that you can't always be

sure that meter scales other than peak will be available on other workstations or plugins if you ever have to use them.

Using The Meters

Thanks to the digital technology that we use today, the way we use level meters has changed over time. If anything, that use has become more relaxed than in the past.

Back in the analog days, it was important for a signal to be recorded and mixed somewhat hot, with levels at 0VU being the goal in most cases (instruments with high transients were recorded lower). The reason was that if a signal was recorded or processed too low, it would be noisy because of the inherent noise floor of analog, which is always there. In other words, if the noise floor of a console was -30VU and we recorded or processed a signal at -20VU, it would only be 10dB above the noise floor, and therefore noisier than if recorded at 0VU and 30dB above the noise floor. It was even worse with magnetic tape, which by nature had a higher noise floor. That meant that in the old analog world, hot levels were your friend.

In the early days of 8 bit digital a similar situation took place. Then it was advisable to record and mix as close to 0dBFS as possible because of a digital type of distortion called quantization noise that increases as the signal level decreased.

Today in our 24 and 32 bit world, that’s no longer a problem, and just about any level will be free of those artifacts. That said, except for the mix buss

(which we'll discuss in a later chapter), most engineers try to keep their meters centered around -10dB with peaks at around -6dB. For most applications, this level allows for sufficient headroom (see the next section) without having to worry about clipping for most program material.

Gain Staging

Gain staging is setting the correct amount of gain for each stage of the signal path so that no one section overloads or clips. On an analog console, that would mean making sure that the input gain doesn't overload the preamp, and the preamp gain is set so it doesn't overload the equalizer section, which has its gain set so it won't overload a outboard device inserted on the channel, which has its gain set so won't overload the fader buffer, which won't overload the master buss. This is the reason why a pre-fader and after-fader listen (PFL and AFL) exist: to monitor different sections of the signal path to make sure that no clipping is occurring.

Setting The Gain

Here are three scenarios in setting the gain that illustrate this procedure in greater detail:

Setting The Gain Using A Signal Present Meter

Set the input level so that the green signal present LED stays lit as long as there is program.

Increase the gain until the red overload LED just lights.

Back off the input level until the red overload LED never lights even on the loudest most transient portions of the signal.

Setting The Gain On An Analog Console

Set the input Trim control so that the peak LED just lights, then back the Trim off so that the LED will not light even on the highest peaks

On a console with an EQ peak LED, set the EQ gain or boost controls so that the LED doesn’t light. If a large amount of boost is required, adjust the input Trim control so that the EQ peak LED does not light even on the loudest sections of the song.

If an outboard device is inserted in the signal path, set its Input control so that its peak LED lights with the occasional peak, then back the control off until the peak LED never lights.

If a channel peak LED exists, set the previous EQ or outboard device Output control so that the peak LED does not light. Adjust the level so that the channel VU meter hits in or around 0VU or a channel peak meter at around -10dB with peaks at around -6dB.

Setting The Gain In A DAW Mixer

Make sure that the channel peak meter is around -10dB with peaks at around -6dB.

If the meter indicates either lower or higher, either insert a Trim plugin or another processor plugin.

Set the plugin Input so the peak LED lights on the loudest sections of the song, then back it off so that it never lights.

If additional plugins are required, set the next one in the signal path so that peak LED lights on the loudest sections of the song, then back it off so it never lights. Do this by adjusting the Input control of the plugin. If the Input control adjustment requires a setting either at the extreme high or low of its range, then back off on the output level using the Output control of the previous plugin.

Adjust the level of the last plugin so that the channel peak meter at around -10dB with peaks at around -6dB.

Setting The Subgroup And Master Fader Level

Many engineers like to group certain instruments together because it's easier to control the level of one fader than many at the same time. Some engineers prefer VCA groups (Voltage Controlled Amplifier) where the faders are all linked together, while other mixers prefer subgroups where the output of the certain individual channels are fed into a dedicated subgroup channel, or even combinations of both.

While all of these stages were slightly tweakable, a rule exists in the analog world that aptly applies to the digital realm as well.

RULE 1

The level of the channel faders should always

stay below the subgroup or master fader.

This means that the level of the master fader should always be placed higher than each of the channel faders (see Figures 4.1, 4.2, and 4.3). While it might be okay if one or two channels is slightly above the master (it's almost inevitable in every mix), just a single channel with big chunks of EQ (like +10 of a frequency band) or an insert with an effects plugin with a level that's maxed can destroy any semblance of a good-sounding mix.

Graphical user interface, application Description automatically generated

Figure 4.1: Channel faders too high, subgroup fader too low.

A picture containing text, row, lined, steel Description automatically generated

Figure 4.2: Subgroup too high, master fader too low.

A screenshot of a computer Description automatically generated with low confidence

Figure 4.3: Channel and subgroup faders at correct levels.

Because many large-format analog consoles have more than sufficient headroom, this rule isn't always religiously followed, but it has been a

golden rule since day one of modern consoles. Not following it is the main reason why many mixes, especially those done in the box, lose fidelity because a submaster or the master buss is overloaded!

Headroom And Why It’s Important

RULE 2

Leave Lots Of Headroom!

Speaking of headroom, this brings us to a second rule:

Recording engineers in the digital world have been taught to increase the levels to as close to 0dB Full Scale as possible in an effort to “use up all the bits.” While this might have been useful back in the days of 16-bit recording, it doesn’t apply as much today in the 24-bit and beyond world. The short reason for this is that if a piece wasn’t recorded hot enough in the 16-bit days, it would end up being noisy as a result. That’s not true in our 24-bit world. Today we can record at a level well below 0dBFS and not have to worry about noise, and there’s a great advantage in doing so: staying in this range gives us plenty of headroom and that means a bigger, more open sound.

Headroom is our friend. We need it to preserve the super-fast transients that make up the first part of the sound of just about any mix element (but especially instruments such as drums and percussion). These transients can

typically range as high as 20dB above what a VU meter might be telling you (peak meters are much closer to the true level), and recording too hot means cutting them off by going into overload if only for a millisecond or two. This results in not only a slightly dull recording but one that sounds less realistic as well. The solution: more headroom.

Headroom means that our average level might be –10dB or less on the meter, leaving plenty of room for transients above that. This concept is actually once again a holdover from the analog days, when a really good console might have a clipping point of +28dB at the master mix buss. Since 0dB VU = +4dB (without getting into an explanation heavy in math), this means that you had a full 24dB of headroom before you ran into distortion as long as you kept your mix hovering around 0VU.

While leaving 24dB for headroom might be excessive in the digital world, leaving 10 or 15dB is quite sufficient. Since it's easy enough to make up the gain later, you won't increase the noise, and your mix will be cleaner, so why not try it?

TIP: When using large amounts of EQ or a plugin with a lot of gain, lower the channel fader or the plugin output rather than bringing up the other faders around it.

The Overall Approach

Whether they know it or not (and many pros aren't aware of how they do it), most great mixers have a method in the way they approach a mix. Although the method can vary depending on the song, the artist, the genre,

or even if the mixer recorded the song from scratch or is just coming in for the mix, the technique remains constant as illustrated below. The last point may be the most important in creating an outstanding mix. As nineteen-time Grammy Award winning mixer Benny Faccone so succinctly states,

1. Determine the Direction of the Song

2. Develop the Groove

3. Find the Most Important Element and Emphasize It

“It’s almost like a musician who picks up a guitar and tries to play. He may have the chart in front of him, but soon he has to go beyond the notes in order to get creative. Same thing with mixing. It’s not just a thing of setting levels any more, but more about trying to get the energy of the song across. Anybody can make the bass or the drums even out.”

Tall, Deep, And Wide

Great mixers mix in three dimensions. They think “tall, deep, and wide,” which means that all the audible frequencies are represented, there’s depth to the mix, and it has some stereo dimension as well (see Figure 4.4).

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Figure 4.4: Tall, deep, and wide

The “tall” dimension is the result of knowing what sounds balanced frequency-wise as a result of having a strong reference point. This reference point can come from experience as an assistant engineer listening to what other engineers and producers do, or simply by comparing your mix to other songs or mixes that you’re very familiar with and appreciate for their fidelity.

Essentially what you’re trying to accomplish here is to make sure that all the frequencies are properly represented.

Usually this means that all of the sparkly, tinkly highs and fat, powerful lows are present in the final mix. Sometimes some mids need to be cut. Other times you might shave extremes from the EQ. Most of the time, clarity is what you aim for, but that might not be the case in every genre of music. Again, experience with mix elements that sound good really helps find your reference point.

The effects or “deep” dimension is achieved by introducing new ambience elements into the mix. This is usually done with reverbs and delays (and sometimes modulation offshoots like flanging and chorusing), but room mics, overheads, natural ambience, and even leakage from other instruments play an equally big part in creating the depth element as well. Even if you’re using samples, all of these effects come into play.

The panning or “wide” dimension is the process of placing an audio element in the soundfield in such a way that it creates a more interesting soundscape, as well as making sure that each mix element is heard more clearly.

The Six Components of a Mix

Every genre of modern music, be it rock, pop, R&B, hip-hop, country, new age, swing, drum and bass, trance, or any other category featuring a strong backbeat, has six main components required for a great mix. They are:

Balance. The volume-level relationship between musical elements.

Frequency range. Having all needed audible frequencies properly represented.

Panorama. Placing a musical element in the soundfield.

Dimension. Adding ambience to a musical element.

Dynamics. Controlling the volume envelope of an individual track or the entire mix.

Interest. Making the mix unique.

Many mixes have only four or five of these components, but all six must be present for a great mix, as they are all equally important.

In some music genres that require simply re-creating an unaltered acoustic event (such as orchestral or jazz or any live concert recording), it’s possible that only the first four components are needed to have a mix be considered great, but dynamics and interest have evolved to become extremely important components even in these genres as modern tastes have evolved.

We’re going to look at each of these mix components in depth in separate upcoming chapters.

The Intangibles Of A Mix

It’s easy to think that getting a good mix is just a matter of pushing up some faders, getting a reasonable balance, adding some effects, and you’re finished. Although that might work for a rough mix, there are still a number of intangibles that are vitally important to achieving a great mix. Awareness is always the first step in learning, so here are some things to consider before you start to move faders around.

The Arrangement

It’s really easy to get caught up in just the audio portion of being an engineer, but unless you seriously consider how the music itself is put together (assuming that music is what you’re mixing, of course), your ultimate product probably won’t sound great no matter how good you are at balancing tracks.

Anyone with a little mixing experience has found that the arrangement is usually the #1 non-audio problem in a mix. In these days of unlimited tracks, it’s all too easy to pile more and more musical elements along with double- and triple-tracks of everything you can think of. You can easily wind up with a hundred tracks to wade through before you know it, and that leaves you with the impossible task of making it sound like something more than a dense wad of audio goo.

A good producer will usually bring some common sense to the arrangement, paring things down to where they’re adding to and subtracting from the dynamics and intensity of the song, but sometimes the producer can also be the one demanding everything but the kitchen sink be thrown in. If that’s the case, if the songwriter doesn’t have an innate sense of arrangement (luckily many do), you’ve got a potential mixing mess on your hands.

That’s why it’s important that the mixing engineer be aware of some basic music-arrangement principles. We’ll cover this in greater detail in Chapter 5, “Mix Balance,” because a big part of being a mixing engineer is knowing when to mute certain music elements and knowing just what elements need to take precedence at a certain part of the song.

Good balance actually starts with a good arrangement, so it’s important to understand musical arrangement because so much of mixing is actually

subtractive by nature. This means that the arrangement, and therefore the balance, is changed by the simple act of muting or lowering the level of a mix element whose part doesn't fit well with another or doesn't enhance the dynamic envelope of the song. If the mix elements fit well together arrangement-wise so that they don't fight one another rhythmically or frequency-wise, then your life as a mixer just became immensely easier.

The Performances

Back in the '50s, '60s, and '70s before recording and production techniques rose to the sophistication of today, a shaky performance by a player or singer could pretty much be accepted in the context of the song. With most players only getting a few takes at most to capture a part, the performance was evaluated more on feel than on precision.

Today's production techniques have evolved to the point where precision and feel are equally in demand during a performance, and as a result, the average listener has evolved to unconsciously expect that as well.

Shaky performances that are out of tune and rhythmically out of time can sometimes work if the mix is incredible, but usually only if the track is rock solid to begin with. One of the reasons why a contemporary mixing engineer has a huge box of tools, such as pitch correction, sound replacement, compression, and copy/paste editing, is to help fix or enhance less than perfect performances so they'll make the mix fit together better. That said, there's a limit to how much a performance can be artificially helped before it sounds unnatural, which negatively impacts the mix as well. Like so many other elements given to the mixer, the better the original performances, the better the mix will sound.

The Point Of Interest

Every song has something that's the main point of interest or something so compelling that you can't take your ears off it. If it doesn't, send the song back to the drawing board. It's not complete.

Although having control over the first five components mentioned earlier may be sufficient for many types of audio jobs and might be just fine to create a decent mix, most popular music requires a mix that can take the song to another level. Although it's always easier with great tracks, solid arrangements, and spectacular playing, a great mix can take tracks that aren't particularly special and transform them into hit material so compelling that people can't hear enough of it. It's been done on some of your favorite all-time songs.

So how can we get to that point?

More than being just technically correct, a mix must be as interesting as a good movie. It must build to a climax while having points of tension and release to keep the listener subconsciously involved. Just as a film looks bigger than life, a great mix must sound bigger than real life. The passion and the emotion must be on a level where the listener is sucked in and forced to listen.

And the way to do that? Find whatever mix element is the most important to the song and build from there. In some cases (such as dance and hip-hop

music), the most important element is the groove. Yet in other genres (such as country), it’s the vocal, but in rock and pop it might be a signature line or hook or even the groove or the rhythm.

Even though the most important element is often the lead vocal, it doesn’t necessarily have to be. It could be a riff, such as the opening guitar from the Rolling Stones’ “Satisfaction,” or the intro to Coldplay’s “Clocks,” or the keyboard riff of Ed Sheeran’s “Shape Of You.” It’s always a part so compelling that it forces you to listen to the song.

The Signs Of An Amateur Mix

Before we can talk about how to create a great mix, it’s good to be aware of the signs of one that isn’t that great. Does your mix have any of these characteristics?

The mix has no contrast. That means that the song has the same musical or sonic texture throughout the entire song, or the mix is at the same intensity through the song. This is often the sign of a poor musical arrangement, but as a mixer, it’s your job to sort this out.

The mix has a wandering focal point. There are holes between lyrics where nothing is brought forward in the mix to hold the listener’s attention, or the focal point keeps on changing to something new every section or sooner.

The mix is noisy. Clicks, hums, extraneous noises, count-offs, and sometimes lip-smacks and breaths can be clearly heard. Remember that noises are cumulative in a mix.

The mix lacks clarity and punch. The instruments aren't distinct, or the low end is either too weak or too big.

The mix sounds distant and devoid of any feeling of intimacy. The mix sounds distant because of too much reverb or overuse of other effects.

The mix has inconsistent levels. Mix element levels vary from balanced to quiet or too loud, or certain lyrics or instrument lines can't be distinguished.

The mix has dull and uninteresting sounds. Generic, dated, or often-heard sounds are being used. There's a difference between using something because it's hip and new and using it because everyone else is using it.

Visualizing The Mix

By and large, most mixers can hear some version of the final product in their heads before they even begin to mix. Sometimes this is a result of countless rough mixes during the course of a project that gradually become polished if the engineer is mixing a project that he or she has tracked. Even

if an engineer is brought in specifically to mix, many won't even begin until they have an idea of where they're going before they start.

Many engineers who can hear the finished product before they begin normally start a mix the same way. They become familiar with the song either through a previous rough mix or by simply putting up all the faders (virtual or otherwise) and listening for a few passes. Sometimes this is harder than it seems, though. In the case of a complex mix with a lot of tracks, the mix engineer may have to spend some time muting mix elements before the song begins to pare down and make sense.

"I always try to have a vision of the mix when I start. Rather than just randomly pushing up faders and saying, 'Well, a little of this EQ or effect might be nice,' I like to have a vision as far as where we're going and what's the perspective."

—Ed Seay

"If I know the song, then I already have a pretty clear picture of what I'd like it to be. If not, I'll usually get that the first time I listen through a track. It's not so much for the sonics, but more in terms of size, like figuring out how big the chorus will be. Sometimes I'll get really specific ideas about effects that I'll try as well. I think the main thing, especially if it's a song I haven't recorded, is that I go through instrument by instrument to see how it sounds, but what I'm really doing is learning every single part so that when I come to build my balance, I know where everything is going to be."

—Andrew Scheps

"Luckily, about half the stuff I do these days I'm producing, where I'm always cutting in mix mode, so it sounds like the record right from the beginning."

—Jon Gass

For better or worse, the engineer's vision might change along the way, thanks to input from the producer and/or artist. Considering how often a mix is done in a personal studio these days, a major mixer frequently completes the job without the producer and/or artist ever coming to the studio, although many mixers actually prefer the input. However, a vast majority would prefer to start the mix by themselves and have the artist come by to offer suggestions five or six hours later, after the mix begins to take shape.

Learning To Visualize The Mix

If you're just starting out mixing, you might think, "How can I hear the final product before I've even begin?" Until you gain enough experience to where you understand the possibilities of the DAW you're working on and its plugins, it's not going to be second nature to you as it is for a mixer who's been doing it for years. Not to worry, all you need are a few questions to help mold your vision a bit, and the way to do that is to go back to the six basic mix elements and ask yourself:

How do I hear the final balance?

How do I hear the mix elements EQed"

How do I hear everything panned?

How do I hear everything compressed?

How do I hear the ambience in the track?

What do I hear as the most interesting mix element in the track?

Even if you can answer these questions, you still may not have a full picture of your final mix, but at least you'll have a general idea, which is the first step to a great mix.

Keep in mind that the producer and musicians have a powerful say in the mix as well, and your version of the mix can suddenly take a wide left turn with their input.

That's okay, because after you've gotten everything to the point where you hear the finished song in your head (or even beyond), a left turn should be easy.

This brings us to the nitty gritty of the book, where all the components of a great mix are detailed even further in the next chapters.

5

Mix Balance:

The Mixing Part Of Mixing

The most basic component of a mix is balance. A great mix must start here first, because without balance, the other mix elements pale in importance. There's more to balance than just adjusting some faders though, as we'll see.

The Arrangement: Where It All Begins

Good balance starts with a good arrangement. It's important to understand arrangement because so much of mixing is subtractive by nature. This means that the arrangement, and therefore the balance, is changed by the simple act of muting an instrument whose part either doesn't fit well with another or doesn't fit in a particular section of a song. If the instruments fit well together arrangement-wise so they help build the song dynamically and don't fight one another frequency-wise, then the mixer's life becomes immensely easier.

Tension And Release

All art is built around tension and release, which is just another expression for contrast. Typically, contrast is big against small, fat against slim, wide against narrow, and black against white. In photography, contrast is reflected in the interplay between light and

shadow. In painting it's depth versus surface. In music it's loud against quiet, and in mixing it's full against sparse. Contrast makes things interesting; you never know how big something is until you see something small to compare it to, and vice versa.

All good arrangements are filled with dynamic changes, which means loud versus quiet and full versus sparse. One of the most critical jobs of a mixer is to create this tension and release when it's not there, and when it is, to emphasize it.

This is done by muting and unmuting tracks and changing the level of certain vocals or mix elements at points within the song.

Conflicting Mix Elements

When two mix elements occupy the same frequency band and play at the same volume at the same time, the result is a fight for attention. Think of it this way: you don't usually hear a lead vocal and a guitar solo at the same time, do you? That's because the human ear isn't able to decide which to listen to and becomes confused and fatigued as a result (see Figure 5.1).

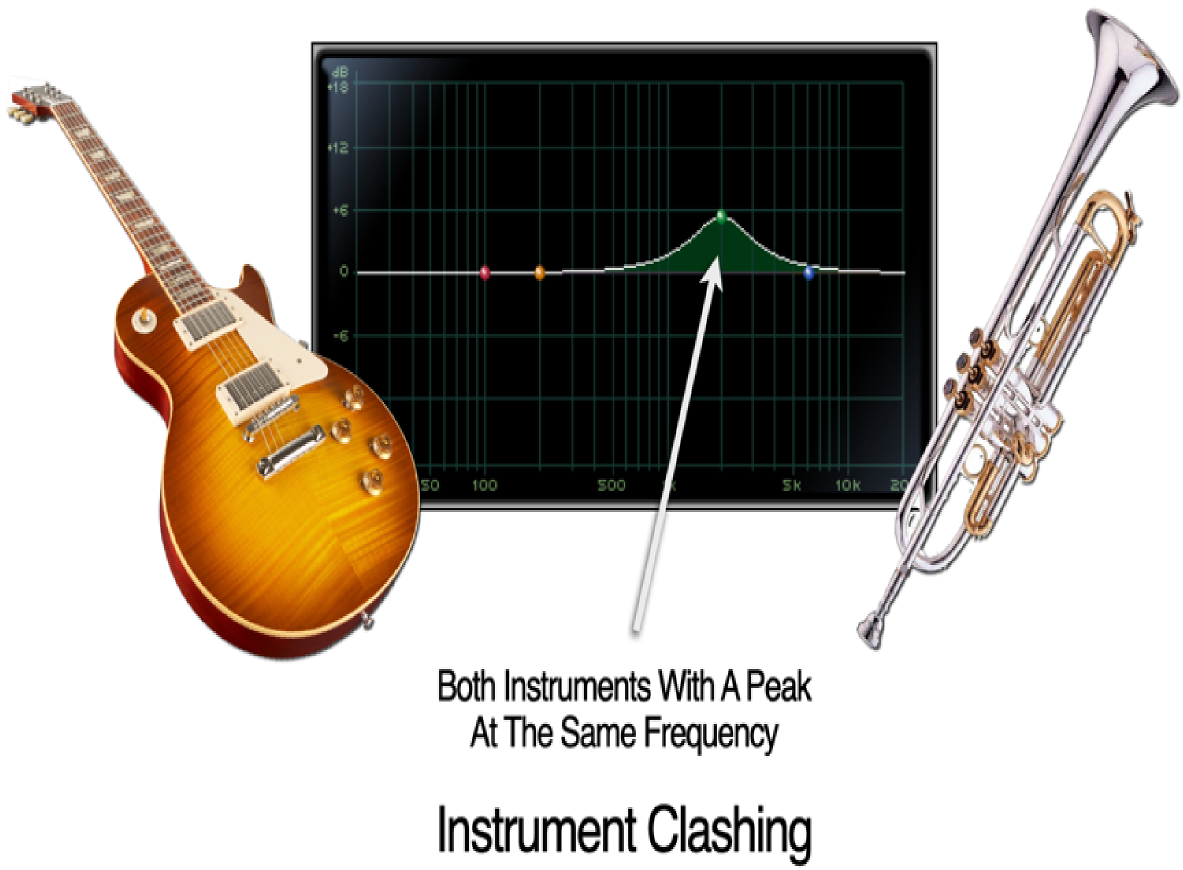


Figure 5.1: Conflicting instruments

So how do you get around instruments conflicting with one another? First and foremost, a well-written arrangement keeps instruments out of each other's way right from the beginning. The best songwriters and arrangers have an innate feel for what will work arrangement-wise, and the result is an arrangement that automatically lays together without much help.

It’s not uncommon to work with an artist, producer, or band that isn’t sure of the arrangement or is into experimenting and just allows an instrument to play throughout the entire song, thereby creating numerous conflicts. This is where the mixer gets a chance to rearrange the track by keeping what works in the mix and muting any other conflicting mix elements. The mixer can influence not only the arrangement this way, but also the dynamics and general development of the song as well.

Arrangement Elements

To understand how arrangement influences balance, we have to understand the mechanics of a well-written arrangement first.

Most well-conceived arrangements are limited in the number of arrangement elements (not mix elements; there’s a difference) that occur at the same time. An arrangement element can consist of a single instrument, such as a lead instrument or a vocal, or it can be a group of instruments, such as the bass and drums, a doubled guitar line, a group of backing vocals, and so on.

Generally, a group of instruments playing exactly the same rhythm can be considered a single arrangement element. For example, a doubled lead guitar or doubled vocal is a single element, as is a lead vocal with two additional harmonies or two drum loops that are playing the same rhythm. However, two lead guitars playing different melody lines or chordal rhythms are two separate elements in the arrangement. A drum loop playing a typical ¼ note dance groove, and a percussion loop playing 8th or 16th note rhythms are two elements. A lead and a rhythm guitar are typically two separate elements as well.

To understand how arrangements work, you have to understand what the basic arrangement elements are first. These may vary a little from genre to genre, but the idea generally stays the same.

The Foundation Element

The foundation is usually comprised of the bass and drums, but can also include a rhythm guitar and/or keyboard if they're playing the same rhythmic figure as the rhythm section. Occasionally, as in the case of rock power trios, the foundation element will only consist of drums, since the bass usually needs to play a different rhythm figure to fill out the band's sound, so it therefore becomes its own element.

The Pad Element

A pad is a long sustaining note or chord that adds a sort of "glue" to the arrangement and therefore the mix. In the days before synthesizers, a Hammond organ provided the best pad, and was later joined by the Fender Rhodes electric piano. Synthesizers now provide the majority of pads, but real strings or a guitar power chord can also serve in that role as well.

The Rhythm Element

The rhythm element can come from any instrument that plays against the foundation element. That can mean a double-time shaker or tambourine, a rhythm guitar strumming on the backbeat, or congas playing a Latin feel. The rhythm element is used to add motion and excitement to the track. Take it away, and the track loses a bit of life and energy.

The Lead Element

A lead vocal, lead instrument, or instrument solo.

The Fills Element

Fills generally occur in the spaces between lead lines or can be a signature line. You can think of a fill element as an answer to the lead. Here's a chart that might explain these elements better (Figure 5.2).

ELEMENT	PURPOSE	TYPICAL INSTRUMENT
FOUNDATION	The instruments that provide the groove or pulse of the song	Drums and bass
PAD	Long sustaining notes that glue the mix elements together	Organ, electric piano, strings, guitar power chords

RHYTHM	The instruments that provide motion to the song	Percussion, rhythm guitar
LEAD	The focal point of the song	Lead vocal, lead or solo instrument
FILL	The instruments that fill in the spaces between the lead phrases	Solo instrument, background vocal

Figure 5.2: The arrangement elements

Arrangement Examples

To help you understand how arrangement elements work together to produce great mixes, here are a few examples of some hit songs and their arrangement elements.

"Shape Of You" by Ed Sheeran

This huge hit features all 5 arrangement elements but like most great songs, a max of 4 are featured at any one time. "Shape Of You" seems sparse, but there's more going on than you might think. The intensity of the song comes from the layers that are built on each arrangement element (like the doubled and high and low octave vocals in the choruses). Parts are added to make the song sound bigger and more intense in the choruses and "Oh I" sections, then taken away so there's ample space for the vocal in the verses.

The five elements of the arrangement look like this:

The foundation. Intro keyboards that play mostly throughout, bass, kick, snare and claps in the B sections (sometimes known as pre-choruses), choruses and bridge.

The pad. Keyboards in “Oh I” sections and outro.

The rhythm. Percussion on the verses, guitar in the second chorus

The fills. The “mmm’s” in the verses.

The lead. Ed’s vocals.

Listen closely to the following places for how the arrangement elements work together perfectly:

⇒ The first half of the verse with just the vocal, pad, and rhythm.

⇒ The first four bars of the second verse with just the vocal and kick drum.

⇒ The first half of the bridge with spoken word, synth bed, and sound effects.

⇒ The first half of the first outro chorus with just vocals and kick drum.

⇒ The last out-chorus vocal, where it breaks down to just the lead and harmony vocals.

"good 4 u" by Olivia Rodrigo

"good 4 u" is somewhat of a throwback to a 90s rock song, but with some modern mix elements and a different song form. It has two different feels between the funkiness of the verses and the rock of the choruses, but that just adds tension and release.

The choruses are interesting in that they're very garage-rock in that the drums, bass and guitars all play the same rhythm, hence they become one arrangement element. The sections also build in intensity (something that all great mixes do) in that the first chorus is just the rhythm section, guitars and lead vocals. On the second chorus low vocal harmonies are added, while the last chorus adds full vocal harmonies, which act as a pad.

The arrangement elements look like this:

The foundation. Bass, electronic drums, electric guitars in chorus.

The pad. Vocal harmonies during the 2nd and 3rd choruses.

The rhythm. Electric guitars in 2nd half of the verses.

The lead. The lead vocal.

The fills. Harmony vocals in the intro and verses. Effects at the end of the verses.

"Blinding Lights" by The Weeknd

Once again we have a rather sparse arrangement built around a repeating kick and snare pattern with the bass outlining the chord changes of the song. The bass synth acts as a pad since the notes are held so long, which is unusual. The other keyboard pad happens during the chorus to build intensity and color.

For the most part, "Blinding Lights" doesn't have many fills, with only vocal answers in the bridge.

The five elements of the arrangement look like this:

The foundation. Kick, snare.

The pad. The bass synth, keyboard in choruses and ending.

The rhythm. The keyboard in the verses.

The lead. As almost always, it's the lead vocal.

The fills. The fills are handled by the background vocals and the occasional percussion sound effect.

"Refugee" by Tom Petty and the Heartbreakers

Here's a hit by a legendary rock band featuring the classic setup of two guitars, bass, drums, keyboards, lead vocals and background vocals. As with most hits, the dynamics in "Refugee" are great, but they're not created by additional overdub layers. What you have here is real dynamic playing by the band, which doesn't generally happen in a studio recording. The band's playing breathes with the song, pushing it to a peak in the bridge and bringing it back down to a quiet third verse.

Let's look at the arrangement elements:

The foundation. As with most songs, the foundation element for "Refugee" is held down by the drums and bass.

The pad. You can't get a better pad element than a Hammond B-3, and that's what you hear here.

The rhythm. There's a shaker that's placed low in the mix so it's not obvious, but it really pushes the song along with a double-time feel, and you'd miss it if it wasn't there.

The lead. Tom Petty's lead vocal and Mike Campbell's tasty guitar in the intro and solo.

The fills. Once again it's the guitar in the verses and the background vocal answers in the chorus.

The song builds and develops in a classic way. In the intro, the full band is playing with the lead guitar. Then, in the verse, it's just the organ and rhythm section with rhythm guitar strums every four bars. The last half of the verse gets louder as the guitar kicks in, and in the chorus, the guitars go back to what they played in the intro (but they're lower in the mix) and the background vocals answer the lead vocal.

The only thing fancy in this song is the doubled lead vocal in the bridge, and the fact that the first half of the solo is by the organ, followed by the guitar.

"Forever After All" by Luke Combs

“Forever After All” is a glowing example of the “new” country music in that it closely resembles layered pop music except for the addition of traditional country instruments like steel guitar (fiddle and banjo, although they’re not used here, are often added today as well). As you would expect from a big-budget act, this song has absolutely state-of-the-art arranging, which is needed for a song with a simple form.

This song is a prefect example of multiple fills that each have different sounds. There’s always a guitar or keyboard to fill a hole after a vocal phrase, but the same sound is rarely heard from more than once. As is the case with big budget hits, you’ll also hear multiple fills even within the same section, sometimes with one answering the other.

The song dynamics follow a simple but well-used formula, where the verses have much less intensity and the choruses sound huge both instrumentally and vocally.

Let’s look at the elements:

The foundation. Bass and drums

The pad. Steel guitar, big electric guitar chords, and organ during the choruses.

The rhythm. Acoustic guitar and a subtle shaker throughout the song.

The lead. Electric guitar in the intro and interlude, lead vocal in the verses and choruses, and lead guitar and steel guitar in the solo.

Fills. Keyboard in the intros, electric guitar in the first and second verses, piano in the second verse.

Rules for Arrangements

There are a couple of easy-to-remember rules that will always make even the densest arrangement manageable.

Limit the number of arrangement elements. Typically, there should not be more than four arrangement elements playing at the same time. Sometimes three elements can work very well, but very rarely will all five elements simultaneously work.

TIP: More than five arrangement elements occurring simultaneously confuses the listener and causes listening fatigue.

Everything in its own frequency range. The arrangement, and therefore the mix, will fit together better if all instruments sit in their own frequency ranges. For instance, if a synthesizer and rhythm guitar play in the same octave, they may clash frequency-wise and fight each other for attention. The solution would be to change the sound of one of the

instruments so they fill different frequency ranges, or have one play in a different octave, or have them play at different times but not together.

"So much of mixing is what you take away, either level-wise or frequency-wise. There are so many things that you have to eliminate in order to make it all sit and work together."

—Former Record Plant chief engineer Lee DeCarlo

TIP: Here are some ways to prevent instruments clashing with one another:

**** Mute one of the offending instruments so that they never both play at the same time.***

**** Lower the level of the offending instrument.***

**** Tailor the EQ so that the offending instrument occupies a different frequency space.***

**** Pan the offending instrument to a different location.***

**** Change the arrangement and re-record the track.***

Building The Mix

Different mixers start from different places when building their mix. This has as much to do with their training and experience as with the type of material being mixed.

For instance, many old-time New York mixers and their protégés start from the bass guitar and build the mix around it. Some work on the vocal first, since it's the most important element of a song. Some mixers work from the drum overheads first, tucking in the other drums as they go along, while others work first from the toms.

Many mixers mix with every channel always in the mix, only soloing specific mix elements that seem to exhibit a problem. Still others are completely arbitrary, changing their starting place from song to song depending upon whatever mix element needs to be focused on. And of course, so many mixers these days begin with the kick drum.

"I start with everything on, and I work on it like that. The reason is that, in my opinion, the vocal is going to be there sooner or later anyway. All the instruments are going to be there sooner or later, so you might as well just get used to it. And I think that's also what helps me see what I need to do within the first passage."

—Jon Gass

"It really is like building a house. You've got to get the foundation of bass and drums and then whatever the most important part of the song is, like the vocalist, and you've got to build around that. I put the bass up first, almost like the foundation part, then the kick in combination with the bass to get the bottom. I build the drums on top of that. After I do the bass and drums, then I get the vocal up and then build everything from there."

—Benny Faccone

"I start with the drums and bass and get the basic foundation, then add the guitars and piano or anything that would be considered part of the rhythm

section. After that feels good, then I put a vocal in, because the style of music that I do is all vocal-driven, so the sooner I get it in the mix, the better. After that, I place any of the ear candy around the vocal and rhythm section."

—Bob Bullock

"I'd love to say that I always build it from the vocal, but usually what I'll do is deal with the drums to get them to act like one fader's worth of stuff instead of 20 or whatever it is. Once I've done that, everything seems to come up at once."

—Andrew Scheps

"It's totally dependent on the music, but if there was a method of my approach, I would say the rhythm section. You usually try to find the motor and then build the car around it."

—Legendary engineer Bruce Swedien

"I typically bring in the overheads first because my overheads are overall drum mics, then I fit the snare into that, then I get the kick happening. At that point I take a look at the toms and see if they're really adding a lot of good stuff to the drum sound. I'll just keep them on and set them where I want and then push them for fills, then tuck in the hat."

—Legendary engineer Allen Sides

There are actually a couple of directions to go when it comes to building a mix and both are valid. The first method is to build the mix by ear, which can be somewhat unrepeatable when it comes to balances.

Sometimes you get it just right, and other times you struggle to find the right combination of levels. Obviously the more experience you have, the better and more consistent you are at doing it.

The second method is designed to get you in the ballpark a lot faster and that's building a mix by using the meters.

Let's look at both.

The Mix By Ear Method

When building a mix by ear, there is no standard mix element to start and build a mix from, although most mixers start with the kick. Modern mixers employ various approaches and they all work, especially in different genres of music. For instance, here are the places from which a mix by ear can be started:

From the Bass

From a loop or sample

From the Kick Drum

From the Snare Drum

From the Drum Overheads

From the Lead Vocal or main instrument

With all of the instruments and vocals in the mix right from the beginning

When mixing a string section, from the highest string (violin) to the lowest (bass)

There are some experienced mixers that just push up all the faders and start to mix with everything in the mix, which is similar to the second way that we'll build a mix. The theory here is that everything will eventually be in the mix anyway, so you might as well start with all the mix elements there as soon as you can. The advantage to this method is that by hearing all the instruments, loops, and vocals, you're able to make an aural space in the mix for everything. If you insert one mix element at a time and tweak it so it sounds great, you begin to run out of space for the lead element (usually the vocal) so it never sits at the right level in the track. When that happens, you might have to go back to the beginning to make sure everything fits together properly.

Wherever you start from, it's a good idea that the lead arrangement element (usually the the vocal) be inserted into the mix as soon as possible. Since the vocal is generally the most important element, it may

use up more frequency space than other supporting mix elements. Many mixers find that it's difficult to find the right level with the rest of the track by waiting until late in the mix to insert the vocal.

Various Mix By Meters Methods

Setting levels by using the meters has been debated from the beginning of mixing time. Some mixers feel that they can get in the ballpark by setting the levels with the master mix meters alone, while others discount any such method out of hand. The fact of the matter is that for those using the meter method, feel and instinct are still a large part of their technique, making it equally as valid as those who rely on instinct alone.

Regardless of the level you begin at, keep watching your master buss meters for overloads as you add each additional mix element to be sure that a clean signal path is maintained.

As with everything else that you read, try out different methods, use what works best for you and throw away the rest.

"I usually start with the bass at about -5 (on a VU meter) and the kick at about -5. The combination of the two, if it's right, should hit about -3 or so. By the time the whole song gets put together and I've used the computer to adjust levels, I've trimmed everything back somewhat. The bass could be hitting -7 if I solo it after it's all done."

—Benny Faccone

"I'll start out with the kick and bass in that area (-7 VU). By the time you put everything else in, it's +3 anyway. At least if you start that low you have room to go."

—Don Smith

"Usually a good place to start is the kick drum at -6 or -7 or so. I'll try to get a bass level that is comparable to that. If it's not exactly comparable on the meter because one's peaking and one's sustaining, I get them to at least sound comparable. That's kind of a good starting place for me."

—Ed Seay

"I'll get the snare drum constantly hitting the back beat of the tune at around 5, and everything gets built around it."

—Lee DeCarlo

TIP: One of the tried-and-true mix level methods is to begin with the first instrument hitting about -10dBFS on a typical digital peak meter. That provides enough headroom so that when additional instruments enter the mix, the master buss meter won't immediately be lighting its overload indicator.

Starting From The Drums

In the early days of recording there was no such thing as balancing the drums since the entire kit was treated as a single instrument and miked using just a single microphone. As producers began to understand how important the beat (and the foundation arrangement element) was, a microphone was added on the kick as well. Eventually the modern drum sound evolved to where each drum and sometimes each cymbal is individually miked. As a result, the internal mix of the drums is a very important part of virtually every modern recording, unless you're using a drum or song loop where this balance is already set.

When mixing by ear, different engineers approach the drum mix in different ways. Some begin with the kick drum and build around that, while others start with the snare or claps, since that provides the backbeat of most songs. Yet others want to build their drum mix around the toms so they don't get lost in the mix, especially if they're prominently featured.

A unique case has the mix being built around the overhead mics (not to be confused with cymbal mics) when they're used. The overhead mics are placed further away from the cymbals than normal cymbal miking because they're meant to pick up the overall sound of the entire drum kit. If overhead mics are recorded, many mixers like to start their mix from there and then fill in the sound with the other drum mics. This won't work so well when the mics are placed lower over the kit (with the idea of just picking up the cymbals) because they won't pick up enough of the rest of the drum kit.

Remember that if you're mixing real drums, the sound of every drum will change anywhere from a little to a lot when a new drum or cymbal is added to the mix due to the leakage from the other drums into the mic.

TIP: Microphone leakage is normal, and although there are ways to attenuate and even eliminate it, sometimes the overall drum sound suffers as a result.

The Modern Mix By Meters Method

The second approach to building a mix is by setting each mix element to a certain level on the master buss meter. This technique was actually created back in the 1960s when mixers only had VU meters on their consoles to reference to, but was refined over time as mixes for multiple musical artists during live radio or TV broadcasts had to be created in a hurry.

TIP: Mixing by meters is not designed to be the final mix, but will get your mix in the ballpark quickly. Remember that you most likely will have to tweak the balances a fair amount if compression and EQ are applied later.

To use this method, first make sure that your meters are set in Peak mode. Then, go to the loudest part of the song. Now watch only the master buss meters (NOTE: In Logic Pro be sure that both the Stereo Out and Master faders are set at 0, meaning the white line on the fader scale).

Start with the Kick track. Set the level so that only the peaks are touching -5dB on the master fader meter. If there are multiple Kick

tracks, then all Kick tracks played together should touch -5dB on the master fader meter.

Set the Snare track so that only the peaks are touching -5dB on the master meter. If there are multiple Snare tracks, then all Snare tracks played together should touch -5dB.

Set the Tom tracks so only the peaks are touching -5dB on the master fader meter. This means the Tom hits only, not the leakage in between.

Set the High Hat track so only the peaks are touching -20dB on the master fader meter.

Set the Overhead or Cymbal tracks so only the peaks are touching -20dB on the master fader meter. This means the cymbal hits only, not the leakage in between.

Set the Room track(s), if any, so only the peaks are touching -30dB on the master fader meter.

Set the Percussion tracks so only the peaks are touching -25dB on the master fader meter. If there are multiple Percussion tracks, then all tracks together should touch -25dB. Note: low frequency percussion like bongos or congas sometimes work better at -20dB.

Set the Bass track(s) so that only the peaks are touching -10dB on the master fader meter. If there are multiple tracks, then all tracks together should touch -10dB. Note: If the Bass track has a high degree of low end content, then -15dB might work better instead.

Set the Vocal track so only the peaks are touching -10dB on the master fader meter. Note: In some genres of music like rock or dance, the vocal might work better at -15dB instead.

Set the Guitar tracks so that only the peaks are touching -20dB on the master fader meter. If there are multiple tracks playing together at the same time, then all tracks together should touch -20dB.

Set the Keyboard tracks so that only the peaks are touching -20dB on the master fader meter. If there are multiple tracks playing together at the same time, then all tracks together should touch -20dB.

Set the Background Vocal tracks so that only the peaks in the song are touching -15dB on the master fader meter. If there are multiple tracks, then all tracks together should touch -15dB.

Set the Vocal Double track (if there is one) so that only the peaks are touching -15dB on the master fader meter.

Set the Horn tracks or Fill tracks so that only the peaks are touching -15dB on the master fader meter. If there are multiple tracks playing together at the same time, then all tracks together should touch -15dB.

Set the Solo track so that only the peaks are touching -15dB on the master fader meter.

TIP: The more precisely you adjust these levels, the more likely the mix will sound balanced when you're finished.

To summarize the method on a chart, here's what the meter levels would look like, complete with all the variables.

Figure 5.3: Building A Mix Via The Meter Method

TRACK	MASTER METER LEVEL	VARIABLES
Kick	-5dB	
Snare	-5dB	
Toms	-5dB	Only when played; not leakage
High Hat	-20dB	
Overheads (Cymbals)	-20dB	Only when played; not

		leakage
Room	-30dB	increase for extra ambiance
Percussion	-25dB	Try low frequency percussion like bongos or congas at -20dB
Bass	-10dB	-15dB if extra bass content
Vocal	-10dB	-15dB for rock, dance or similar genres
Background Vocals	-15dB	All background vocal tracks together
Doubled Vocal	-15dB	Set 5dB less than lead vocal
Guitars	-20dB	All guitars tracks together
Keys	-20dB	All keyboards tracks together
Horns	-15dB	All horn tracks together
Claps	-15dB	All clap tracks together
Solo	-15dB	Set to -10dB if too low in the mix

TIP: The Meter Method only works with Sample Peak meters, which are usually the default setting of most DAWs. If the meters are set to any other ballistic measurement, then the mix will be unbalanced.

Setting The Right Levels

In the Meter Method we start with the kick drum at -5dB. This is because we are mixing to very precise points on the master buss meter and if those levels are observed, the overload indicators on the master buss shouldn't light despite the high level.

In the By Ear method we start with the kick at -10dB. That's because you're more likely to make one or more tracks louder than what might ultimately be required and cause an overload indicator to lite, so we start with a lower level to compensate.

Vocal First

Wherever you start from, it's generally agreed that the vocal, or whatever is the most prominent or significant melody mix element, should be added to the mix as soon as possible.

The reason for this is twofold. First of all, the vocal is probably going to be the most important element, so it will need to take up some of the frequency space that may already be assigned to other supporting

instruments. If you wait until late in the mix to add in the vocal, there may not be enough space left, and the vocal will never sit right with the rest of the track. You'll find that it will be either too loud or too quiet, and you might not be able to find the right level.

The second reason has to do with effects. If you tailor all of your effects to the rhythm section and supporting instruments, there may be none left when it's time to add in the vocal or most prominent instrument.

What's The Genre?

The type of program being mixed will frequently make a difference on where you build your mix from. For instance, when mixing dance music where the kick is everything, that's the obvious choice for a starting point. Mixing an orchestra, however, requires a different technique.

According to mixer Don Hahn, who mixed hundreds of orchestral sessions for records, movies and television, "The approach is totally different because there's no rhythm section, so you shoot for a nice roomy orchestral sound and make it as big a sound as you can get with the amount of musicians you have. You start with violins, then violas if you have them, cellos, then basses. You get all that happening and then add woodwinds, French horns, trombones, trumpets, and then percussion."

In jazz, the melody instrument can be the starting point with the bass inserted shortly after to solidify the foundation. Early pop and rock engineers started with the bass and built around that, although that approach isn't used much anymore.

The Yardstick Mix

If you’re really happy with your mix but have to send it off to someone else to work on, how do you make sure that they’ll get a mix with the same balance? Assuming that they’ll be using the same plugins (although you could burn the processing into the mix), the answer is what’s known as a “Yardstick Mix.”

A Yardstick Mix is one that you can set all the channel faders to 0 and the mix comes back perfectly balanced just the way you left it. On a console you can literally put a yardstick across the faders to make sure that all the faders at at exactly at the same place. The 0 level point is what’s normally chosen, although just about any level will work as long as your headroom remains intact.

How It’s Done

This type of mix was performed mostly in the console days, especially when recording a live concert. In order to preserve the mix for the next engineer down the line (especially if the ultimate distribution was television or film), you’d use the Yardstick process.

The way it would work is that you’d never change the channel fader level. All of the level adjustment comes from the input gain trim so that the channel fader never has to leave the 0 point.

On a DAW you'd either use the built-in channel trim (if your workstation features one), a dedicated trim plugin, or even the input and output controls on an EQ or similar processor to set the desired level while the fader is at zero.

While the Yardstick Mix wasn't used that often even back in the analog console days, it's used even less today given the instant recall we have with when a DAW session opens, so it's more of a curiosity of the past. Still, it's an interesting approach to be aware of. Who knows, some day you actually might put it to good use.

Mixing With Pink Noise

An interesting mixing method that can be used for quick rough mixes or as a balance starting point for a mix is by using pink noise. Since the frequency response of pink noise is similar to that of the human ear, it can be used as a reference point against a mix.

There are two ways this can be applied:

Method 1: Track Solo Against Pink Noise

Insert a signal generator on your master buss and set it to pink noise.

Set the level to-10dBFS (set to -6dBFS if you think you need more level), or 0VU if on a console with VU meters.

Bring the level of an individual track up until you can just hear it.

Decrease the level of that track until it just disappears.

Repeat with all other tracks.

This method takes a little bit of practice to figure out where to stop when you're decrease the level of the track below the noise. Remember that this will never get you the perfect mix; it will only get you in the ballpark.

This method works very well for things like setting the levels of multiple-miked instruments, like the top and bottom snare mics, different tom levels, in and out kick mics, multiple mics on bass, guitar amp or piano.

Method 2: Pink Noise Mix Layover

Sometimes just inserting pink noise across your mix when its near completion will show that there's a mix element that's too loud. Here's

how to do it.

Insert a single generator across your master buss and set it to pink noise.

Bring the level up until the pink noise is louder than your mix, then back it off a little.

If any individual tracks stick out above the pink noise, they may be too loud in the mix.

Decrease the level of that track until it just disappears.

Repeat with all other tracks if there are any.

TIP: White noise contains equal energy across the frequency spectrum while pink noise has a 3dB per octave rolloff starting at 2kHz that's closer to the way we hear.

The Difference Between Noise Colors

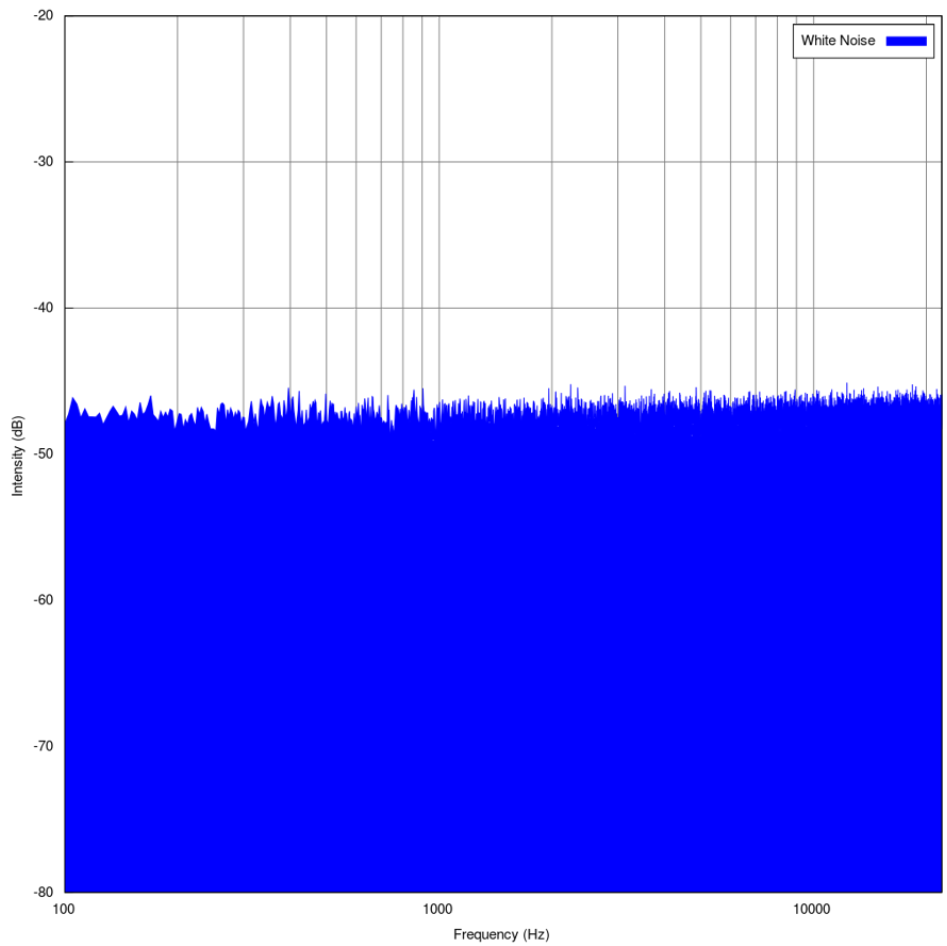
There are more colors of noise than just the standard white and pink noise that are usually readily available in our workstations. Believe it or not, blue and brown noise are also a real thing and used more often than you think. So what are the differences between white, pink, blue and brown noise? It all comes down to frequency and amplitude.

What Is Noise?

While we’re all familiar with background or ambient noise, audio engineers have been dealing with electronic noise for decades. To put it simply, electronic noise (also called “thermal noise”) is the sound of electrons bouncing through electronic components (called “thermal noise”). We hear this noise in the form of hiss, and since the beginning of audio recording, equipment designers have been doing everything they can to keep it to a minimum. That’s not the kind of noise that we’re talking about though.

When it comes to white, pink, blue or brown noises, these are continuous, unchanging, random, uncorrelated samples that are intentionally generated. Originally this was for various kinds of system measurement, but other uses have been found along the way as we’ll see here.

White Noise



White is generated noise with equal amplitude that's distributed evenly across our spectrum of hearing and a little beyond, from 20Hz to 20kHz. It's pretty harsh sounding and doesn't represent too much of what we hear in nature, but it's been found to be extremely useful for one thing – its masking effect.

Masking is when one sound is dominant enough that you concentrate on it rather than other sounds that might be present. Many people use it to help them sleep, or it's sometimes used in an office setting to help workers concentrate better by masking out the sounds of other workers.

Pink Noise

Chart, histogram Description automatically generated

Pink is used most often in audio testing because it closely approximates how we hear. The energy rolls off at about 3dB per octave, and the acoustic energy is equal across all frequency bands. In other words, pink noise is white noise with decreased high frequencies. Since our hearing is more sensitive to higher frequencies than lower ones, this makes the spectrum of pink noise excellent for acoustic tests like room tuning.

Pink noise has recently found a new use in sleep enhancement as well. It was recently discovered that people who fall asleep to pink noise rather than white tend to sleep longer and deeper.

Brown Noise

Histogram Description automatically generated with medium confidence

Brown (sometimes call Brownian or red noise) doesn't actually come from a color but from the person that discovered it – Dr. Robert Brown. It describes the random motion of water in a stream or the incoherent path of a drunk person walking. In this case, the energy falls off at 6dB per octave, which means that there's no energy over 2kHz.

Although brown noise is also used for sleep enhancement, it's actually been a staple of sound designers for sci-fi films for years, often used as background ambience for scenes set in spacecraft engine rooms.

Blue Noise

Chart, histogram Description automatically generated

Blue is the opposite of pink noise in that the level actually increases by 3dB per octave and has minimal low frequency components. That makes it particularly harsh sounding, but it is useful for dithering (decreasing distortion of low level digital signals). Blue noise does have outside analogs though. Retinal cells that are arranged in a blue-noise-like pattern yield good visual resolution, and Checkov radiation (the blue glow of an underwater nuclear reactor) is in a blue noise pattern as well.

Believe it or not, there are other types of noise as well. Violet, grey, green, black and noisy white all have their places in communications, computer or display electronics. However, when it comes to audio, there's only one color that rules and that's pink.

Reference Mixes

As we discussed back in Chapter 3, one way to make sure that your mix stays on track is to constantly refer to one or more reference songs along the way. This allows you to not only check for element volume balances, but also frequency balance against a reference that you know sounds great.

An easy way to do that is with one of comparison plugins like Adaptr Audio Metric AB (see Figure 5.8), which not only allows you to instantly compare your mix with up to 16 other pre-loaded tracks, but perfectly match the level between them all and your mix, as well as pick a specific section of the song to compare it too. Metric AB now has a suite of

measurement tools as well, including a spectrum analyzer, and meters for loudness, dynamic range, phase, and stereo width.

Similar plugins include Nugen Audio A|B Assist and Melda Production MCompare.

Regardless of which one you use, these are invaluable tools for not only discovering how your mix compares to others, but learning how to mix as well.

Graphical user interface, chart Description automatically generated

Figure 5.8: Adaptr Audio Metric AB

Courtesy of Plugin Alliance

The Panorama Element: Placing Mix Elements In The Soundfield

One of the most overlooked (or at least taken for granted) elements in mixing is panorama, which is the technique of locating a sound element within the soundfield. To understand panorama, or "panning" as most of us call it, we must first understand that the stereo sound system — which is two separate audio channels, each with its own speaker — is designed to represent sound spatially. Panning lets us determine where in the listening environment we place any sound.

But panning does more than just that. Panning can create excitement by adding movement to the track and adding clarity to an instrument by moving it out of the way of other sounds that may be clashing with it. Correct panning of a track can also make a recording sound bigger, wider, or deeper, sometimes all at the same time.

The Stereo Picture

What is the proper way to pan a mix element? Are there any rules? Just like so many other aspects to professional mixing, there's a method to follow and this method has evolved over the years with some good reasoning behind it. Nevertheless, it may seem that panning decisions can be pretty arbitrary.

Imagine you're watching a Western movie; the scene is a wide expanse of Arizona desert and right in the middle of the screen sits a cowboy on his horse. Suddenly, a half dozen cattle rustlers begin attacking him

from behind. Unfortunately, we can't see them very well because the cowboy is situated directly between us and them, blocking our view. If they remain stacked behind the main character and the camera doesn't adjust to include them in the shot, their actions are unclear and any dramatic impact is limited (not to mention the fact that whatever money it cost to include six outlaws on horseback in this scene was a complete waste).

Wouldn't it be better if the director had moved the rustlers from behind the cowboy? Or better yet, spread them out across the screen from left and right so the attack feels more immediate and more threatening?

Of course, that's what we do with the pan pot. It gives the mixer (the director) the ability to move the background vocals (the rustlers) out of the way of the lead vocal (the cowboy) so we can hear (see) each element of the track (shot) much more distinctly.

If we imagine that scene playing out in a recording studio instead of a dusty mesa, it might look like this (figure 6.1):

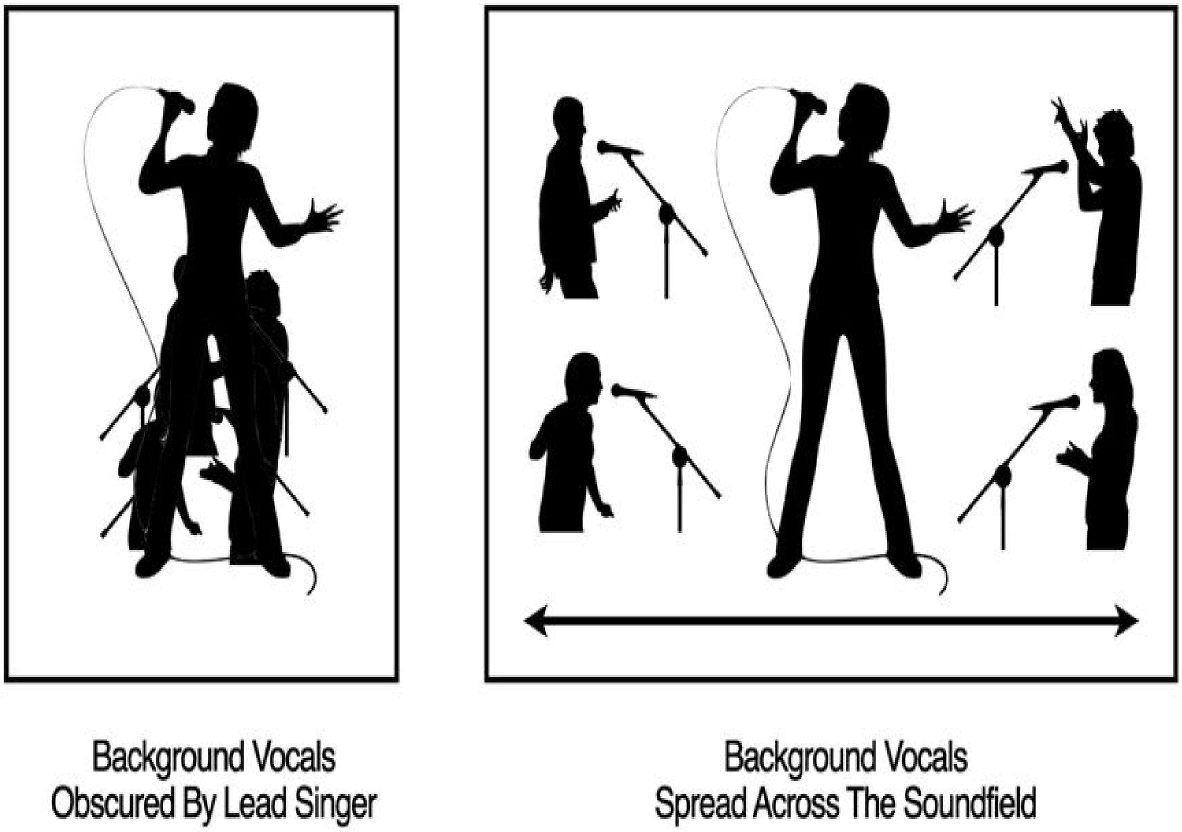


Figure 6.1: Panning visualization.

The Phantom Center

Stereo was invented in 1931 by Alan Blumlein at EMI Records (the patent wasn't renewed in 1959 when the new format was really taking off—d'oh!). Early in its development, a unique and significant feature of stereophonic recording emerged that continues to shape recorded music to this day: a phenomenon known as the phantom center. The phantom

center describes how the output of the two speakers combine to provide an imaginary speaker somewhere in between them (see Figure 6.2).

This imaginary image can sometimes shift as the balance of the music shifts from side to side, which can be very disconcerting to the listener, especially if the speakers are placed far apart. As a result, film sound has always relied upon a third speaker channel in the center to keep the sound anchored. This format is called LCR, for Left, Center, Right, and it became a staple of multiplex movie theaters and high-end home video setups where the center speaker is located near or behind the screen. This third channel never caught on in consumer music circles however, mostly because people had a hard time finding a place for two speakers, let alone three.

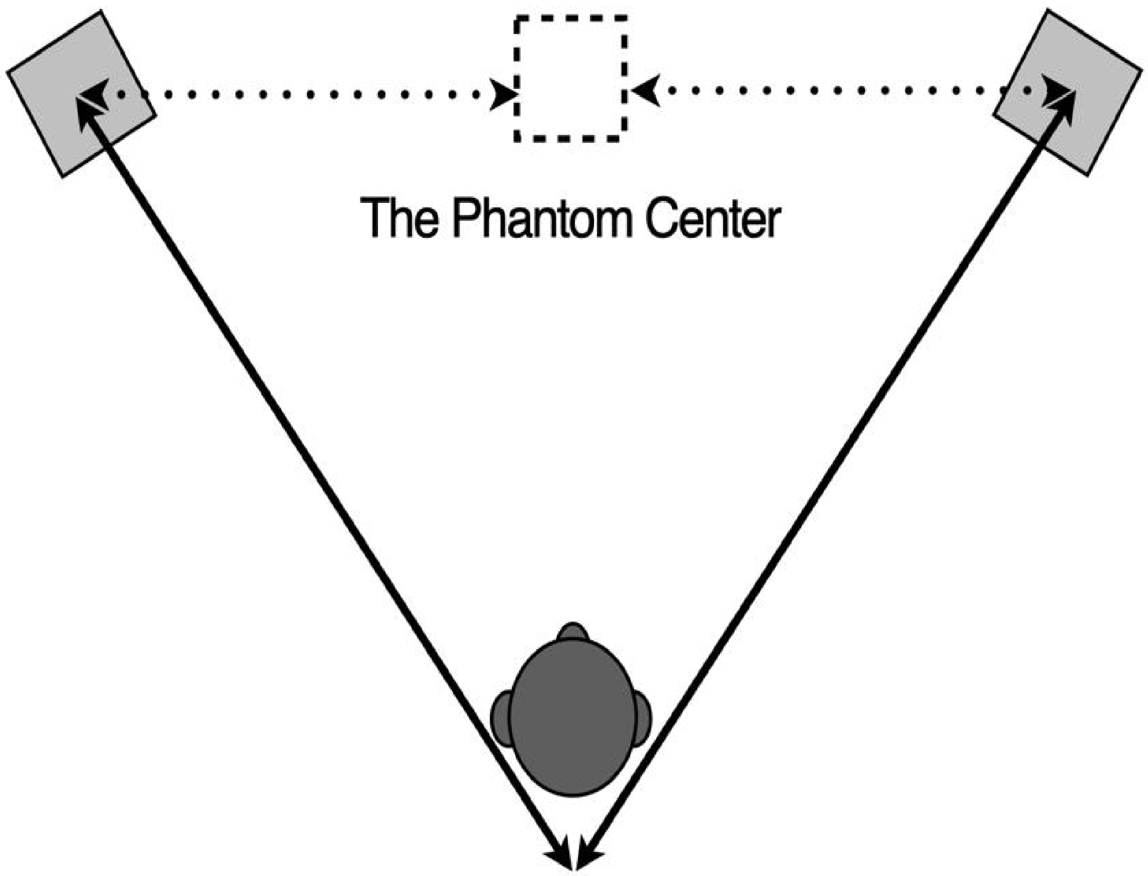


Figure 6.2: The phantom center.

The Big Three

That said, there are three major panoramic areas in a traditional mix: the extreme hard left, the extreme hard right, and the center (see Figure 6.3).

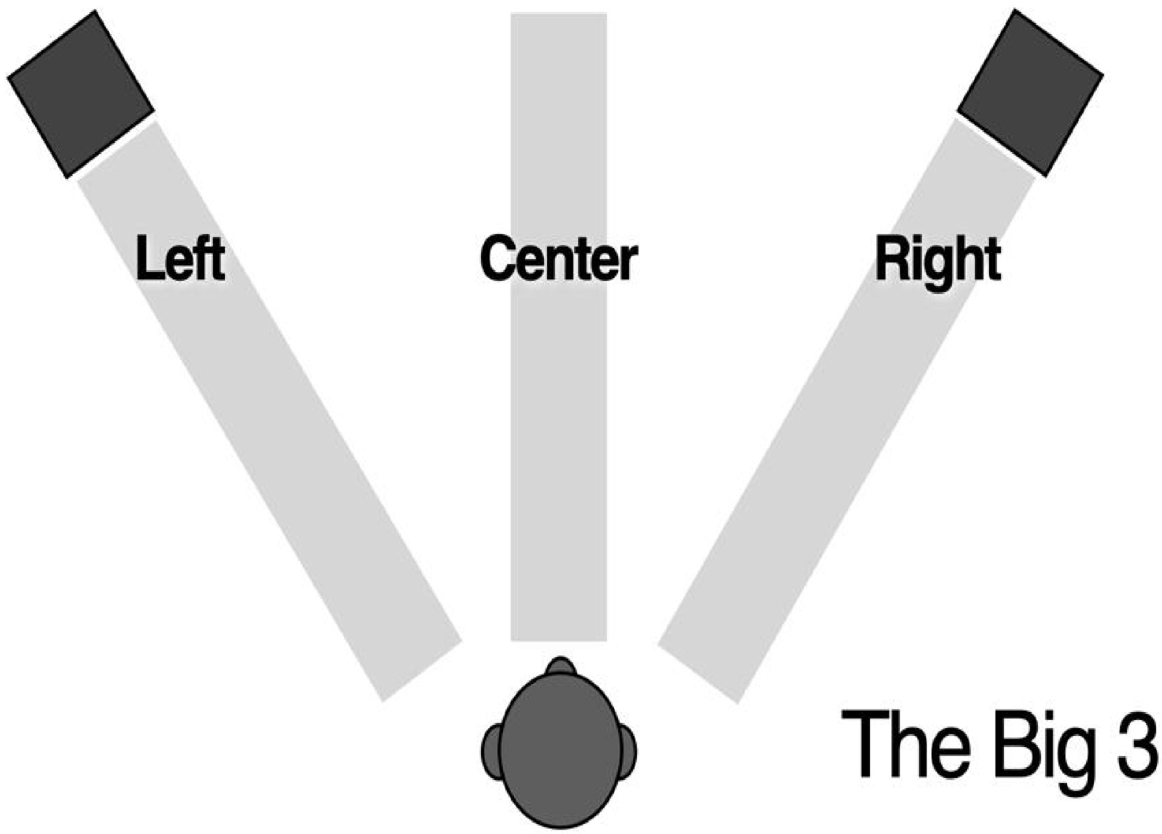


Figure 6.3: The big three panning areas.

The dominant center is an obvious choice to locate the most prominent music element (like the lead vocal) but also the kick drum, bass guitar and even the snare drum. While putting the bass and kick "up the middle" makes for a musically coherent and generally accepted mix technique, its origins date back to the era of vinyl records.

With the earliest stereo recordings in the mid-'60s, the music from the band's instruments was sometimes panned to one channel while the vocals were panned to the opposite channel. This was because stereo was such a new technology, the recording and mixing techniques for the format hadn't been refined yet and what we now know as pan pots weren't available on mixing consoles of the time. Instead, a three-way switch was used to assign the track to the left output, the right output, or both (the center). Sometimes, the channels were even dedicated to either left or right, like on the famous REDD console at Abbey Road Studios that was used to record The Beatles and other major acts of the day.

Because music elements tended to be hard-panned to one side or the other, this caused major disc-cutting problems while mastering. If any low-frequency boost was added to the music on just that channel, the imbalance in low-frequency energy would cause the cutting stylus to literally cut right through the groove wall on the the master lacquer disc (from which all subsequent records were eventually pressed). The only way around this was to either: a) decrease the amount of low-frequency energy from the music to balance the two sides; or b) pan the bass, kick and any other instrument with a lot of low-frequency information back to the center. In fact, a special equalizer called an Elliptical EQ was used during disc mastering to specifically move all the low-frequency energy from both sides to the center in order to prevent any physical cutting problems.

Likewise, as a result of the vast array of stereo and pseudo-stereo sources and effects that have come on the market over the years, mixers began to pan these sources hard left and right almost without thinking, since a mixer's main task is to make things sound bigger and wider. Suddenly things sounded huge! This caused new problems as almost all keyboards and effects devices came with stereo outputs (many are actually pseudo-stereo with one side just chorused a little sharp and then flat against the dry signal) so the temptation rose to pan these new

“stereo” sources hard left and right, one on top of another. The result came to be called “Big Mono.”

“I think that there are three sacred territories in a mix that if you put something there, you’ve got to have an incredibly good reason. That’s extreme left, the center, and extreme right. I’ve noticed that some mixers will get stereo tracks from synthesizers and effects, and they just instinctively pan them hard left and hard right. What they end up with is these big train wrecks out on the ends of the stereo spectrum. Then they pan their kick, snare, bass, and vocals center, and you’ve got all this stuff stacked on top of each other. If it were a visual, you wouldn’t be able to see the things behind the things in front.”

—Dave Pensado

Big Mono

Big Mono occurs when you have a few stereo tracks that are all panned hard right and hard left. In this case, you’re not creating much of a panorama because everything is placed hard left and right so you’re robbing the track of definition and depth because all of these tracks are panned on top of one another (see Figure 6.4).

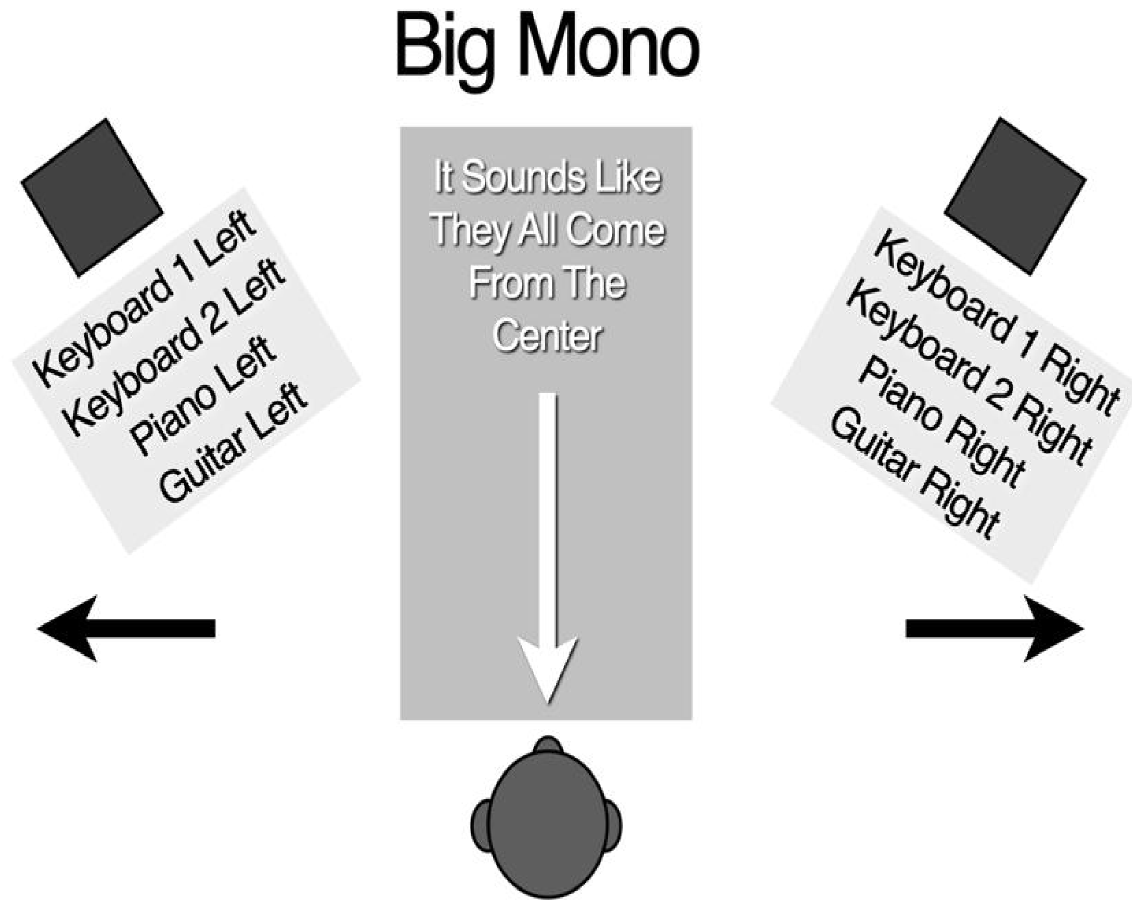


Figure 6.4: Big Mono

"One of the things I don't like is what I call 'Big Mono,' where there's no difference in the left and the right other than a little warble. If you pan the left and right wide, and then here comes another keyboard and you pan that left and right wide, and then there's the two guitars and you pan them left and right wide, by the time you get all this stuff left and right wide, there's really no stereo in the sound. It's like having a big mono record, and it's just not really aurally gratifying. So to me, it's better to have some segregation, and that's one of the ways I try to make everything heard in the mixes. Give everybody a place on the stage."

—Ed Seay

The solution to Big Mono is to not pan anything to the hard left or hard right unless there's a good reason, then don't pan mix elements on top of one another. Instead of panning the two channels hard left and right, find a place somewhere inside those extremes (we'll specifically discuss this in an upcoming section.)

Another is to throw away the effected side of the stereo track (keep the dry channel) if it wasn't truly recorded in stereo and make your own custom stereo patch with either a pitch shifter or a delay (See Chapter 9, "The Dimension Element," for how this is done.)

TIP: Panning narrowly on the same side (like 1:30 and 3:00) provides the width and depth of stereo while still having precise localization within the stereo soundfield.

Panning To The Holes

Another way to think of panning is to make sure that each mix element is placed in its own space within the stereo picture. That means if two mix elements are panned to the left like in Figure 6.5, then it's best to pan the next mix element somewhere in the hole between them.

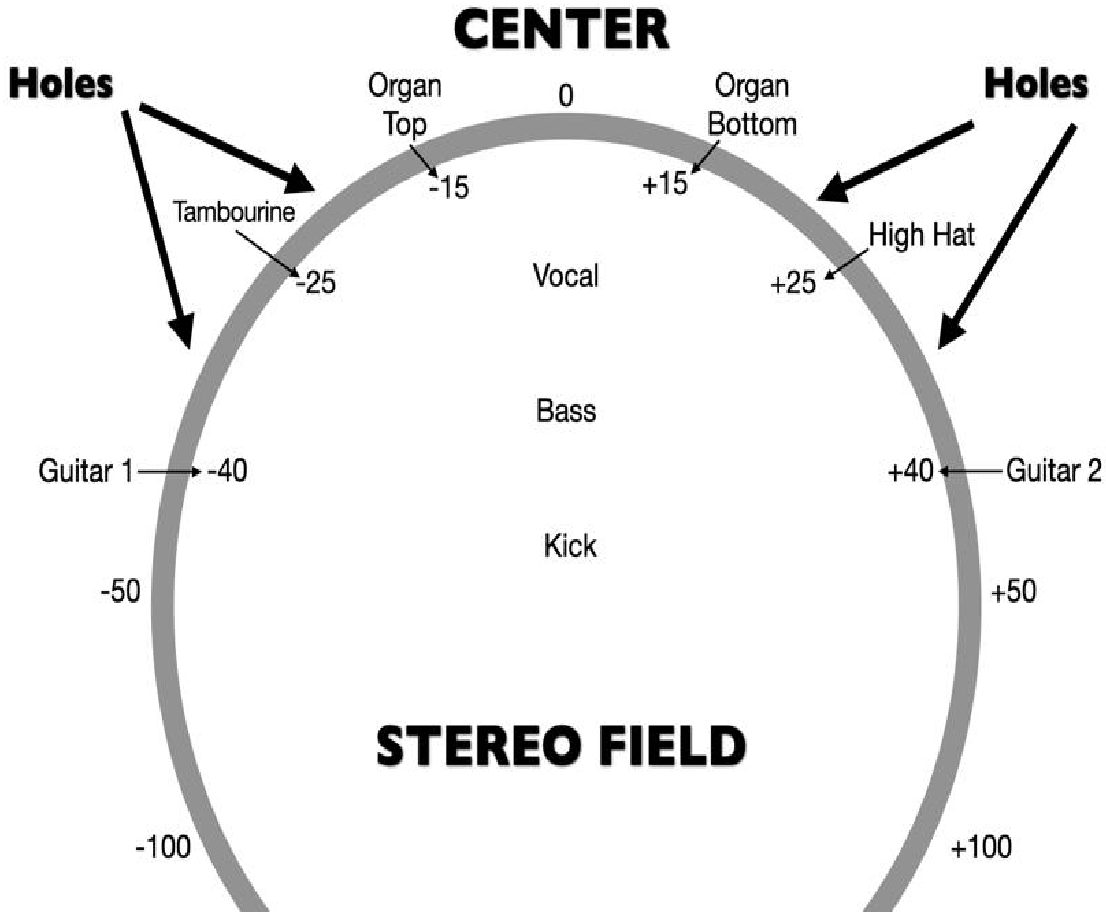


Figure 6.5: Panning to the holes.

In Figure 6.4, the organ Leslie top track is panned to -15 on the pan pot and the tambourine is panned to -25. That means there's a space at -20 where you can pan another mix element. If you were to pan it on top of one of the mix elements at -15 or -25, then you run the risk of one mix element masking the other, which means you'd probably need to EQ one of them to hear them both better. Just panning in the hole eliminates that step.

Likewise, there are holes between -25 and -40, +15 and +25, and +25 and +40 which are wide enough to fit additional mix elements as needed.

TIP: Sometimes you only need a pan separation of a few degrees to eliminate the masking effect between tracks.

Stereo Track Panning Strategy

As you can see, panning stereo mix elements hard left and hard right can actually reduce the spaciousness of what was recorded, but there's another approach to panning that can work better. This method will provide plenty of stereo width while focusing the mix element in the stereo spectrum.

Let's Not Pan Too Wide

While it seems instinctual to want to take a stereo source like overheads or keyboards and automatically pan them hard left and right, keep in mind that most mixers rarely do that (see the quotes from Dave Pensado and Ed Seay, above). In fact most mixers like to keep their maximum width at the 9 and 3 o'clock positions so things never feel too wide yet still keep the dimensional feel of stereo. One exception to this approach is with effects returns, which are frequently panned hard left and right.

Some mixers make the stereo spread even tighter, believing that the closer to the center the mix elements are, the more focused and powerful

the resulting mix is. Regardless of how you feel about sending things out wide, it's best to keep your panning under control except in rare circumstances (see "Panning Outside The Speakers", below).

Stereo With Focus

Despite what you might think, you can achieve a rich, spacious stereo field without panning too wide. Sometimes as little as 10 degrees off center left and right can widen out a mix element in a very discernible way. What's more, you can attain all the focus of a mono track placed in the soundfield while still making it feel wide. There are two ways to do this and they're both illustrated in Figure 6.6.

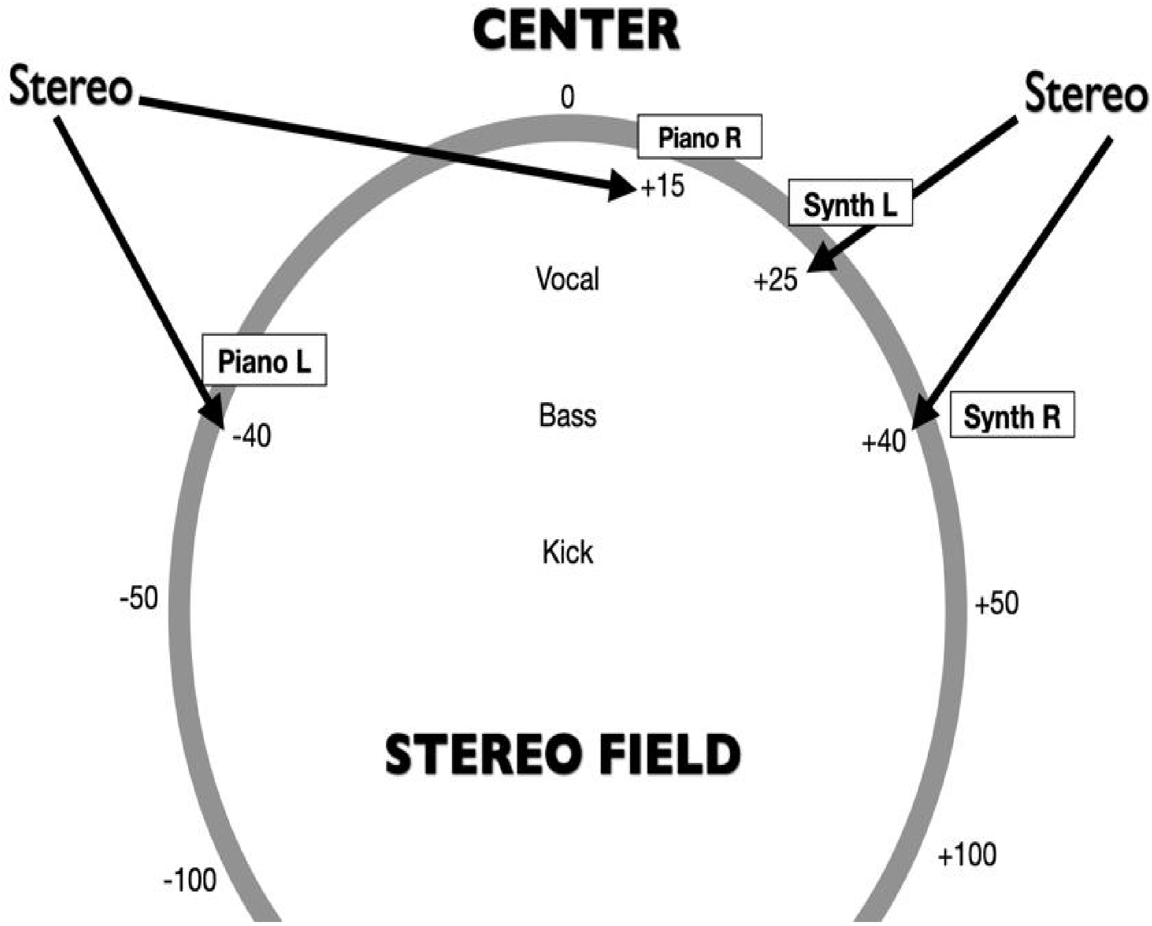


Figure 6.6: Stereo With Focus

On the left side of this mix, you'll see a stereo piano that has its left channel panned to the left at about -40 on the panner and the right channel set to about +15. This provides a feeling of the instrument being on both sides of the soundstage, but it's still firmly focused on the left.

On the right side, the stereo synthesizer is panned between +25 and +40 to the right. Again, this feels wider and deeper than a mono track but

still remains both localized and specific, so it doesn't get lost in the mix.

The idea here, as with mono tracks, is to keep each mix element in its own space across the stereo stage. This will enable it to be heard more cleanly since it won't be masked by another mix element and less processing will be needed as a result.

Panning Outside the Speakers

Some mixers like to use the phantom images afforded by effects processing to pan an instrument outside the speakers. In this case, the phase differences make the instrument seem to come from outside the speakers instead of from them. While some find this effect disconcerting, it can be very effective under the right circumstances.

There are two ways to accomplish this:

On a stereo instrument, flip the phase of one channel. The panning will now seem to expand beyond the speakers.

On a stereo instrument, feed some of the right-channel signal to the left channel out of phase, and feed some of the left-channel signal to the right channel out of phase. This can be done easily by copying both channels, flipping the phase on both, panning them opposite of the way they're already panned, and gradually increasing the level (see Figure 6.7). As the level increases, the sound will seem to pan outside the speakers.

Or, use one of the many plugins that offers a Mid/Side or M/S feature which can provide even more control than the manual method. (check out the Brainworx products, which all have this feature)

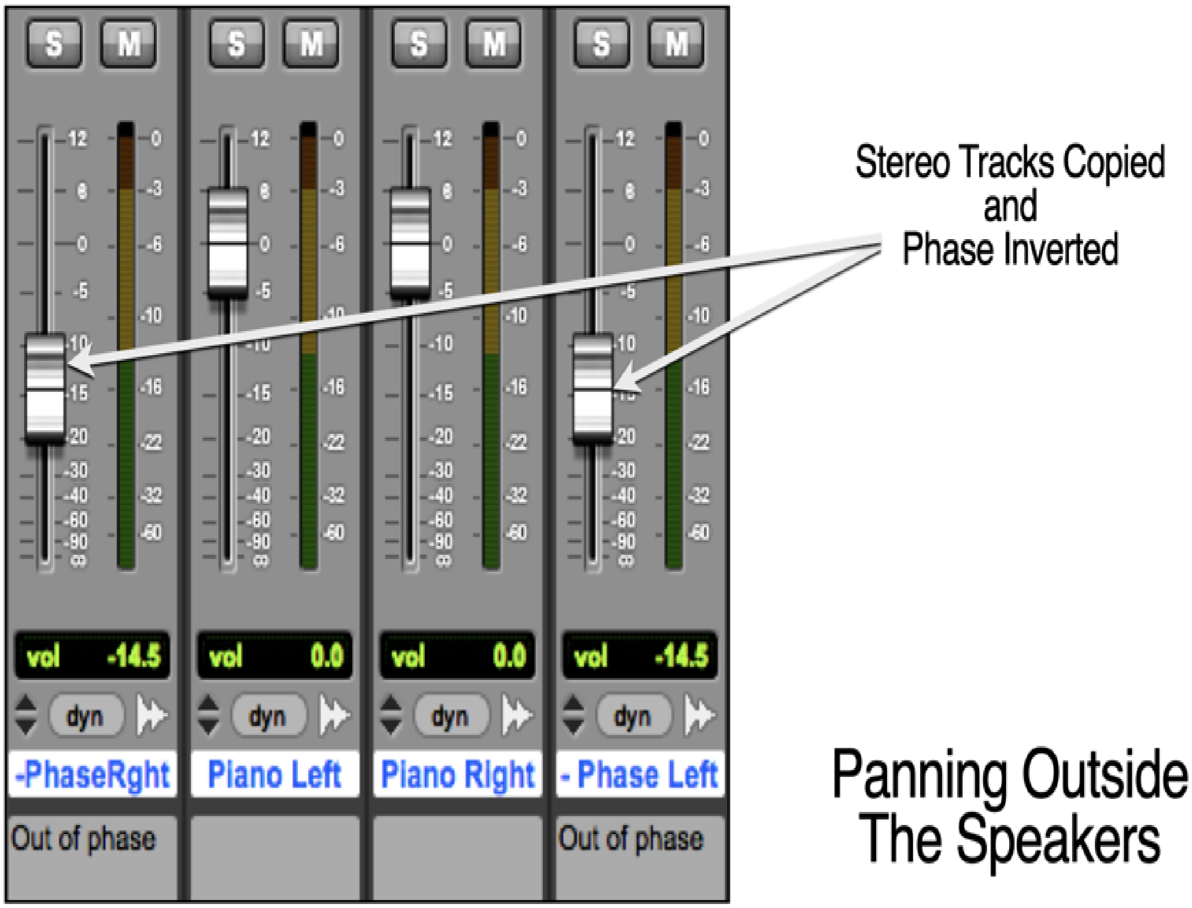


Figure 6.7: Setup for panning outside the speakers.

Beyond Panning For Placement

While panorama usually relates only to placement of a sound source within the stereo soundfield, there are other areas where panning skills can really help a mixer.

Some mixers find that after their panning placements have been determined and all their EQing is complete, just moving the instrument's pan slightly can sometimes provide more clarity — more pop — to the mix. In fact, if you feel that mix elements are fighting for frequency space, the first thing to try is alternate panning.

Some mixers even do some of their panning in mono (yes, that's right!). This is because it's easy to hear differences in phase, which make the instrument stand out when you go back to stereo.

"I check my panning in mono with one speaker, believe it or not. When you pan around in mono, all of a sudden you'll find that it's coming through now and you've found the space for it. If I want to find a place for the hi-hat, for instance, sometimes I'll go to mono and pan it around, and you'll find that it's really present all of a sudden, and that's the spot."

—Don Smith

TIP: When mixing a dance club record, it's best not to pan important elements very wide, since the speakers can be on different sides of the dance floor so half the people won't hear it. It's best to keep important elements either up the middle or maybe at 10:30 and 1:30.

Immersive Audio: Beyond The Stereo Soundfield

To many people, immersive audio seems like something new, but the underlying technology has been around since the earliest days of theatrical motion pictures. The film industry has been using a variety of immersive formats for almost a hundred years in an effort to enhance the experience of the theater audience. While the formats and production gear have gotten more complex with each new format, the idea has always been to provide a "you are there" experience for the viewer. By which, of course, I mean the listener…

A Bit of History

As we now know, immersive audio (or "Surround Sound" as it was formerly known) has been with us in one form or another way longer than you might think. Theatrical films began using the three-channel "curtain of sound" developed by Bell Labs back the early 1930s when it was discovered that a dedicated center channel provided the significant benefit of anchoring the center by eliminating phantom images that wandered around the theater (in stereo, the center image shifts as you move around the room). This also provided better frequency response matching across the soundfield — an added byproduct.

The addition of a rear effects channel to the front three channels dates as far back as 1941 with the "Fantasound", a four-channel system utilized by Disney for the ground-breaking animation classic, Fantasia. It emerged again in the 1950s with Fox's Cinemascope (Do you remember the film "The Robe"? Me, neither.) But rear-channel applications didn't come into widespread use until the 1960s, when Dolby Stereo became the de facto

surround process for all high-end motion picture releases. This popular format uses four channels (left, center, right, and a mono surround, sometimes called LCRS) and is encoded onto two tracks.

Until recently, Dolby Stereo was the standard delivery format for all major films produced for theatrical release and broadcast television as it had the added advantage of playing back properly in stereo or in mono if no decoder is present. This flexibility meant that the original Dolby format was compatible with a wide variety of both new and old theater sound systems. Today it serves mostly as a backup for theaters still exhibiting movies on 35mm film.

In the 1980s, the emergence of digital delivery formats capable of supplying more channels saw the number of rear-surround channels increase to two, and a low-frequency effects channel was introduced to create the six-channel format “5.1”, which soon became the modern standard for most films, surround music, and digital television. Today, we’ve graduated to even more advanced formats such as 7.1, 11.2, and the revolutionary multi-speaker Dolby Atmos system, which debuted in 2012.

And then there was the four-channel Quad from the 1970s, the music industry’s attempt at multichannel music that killed itself as a result of two non-compatible competing standards, both of which suffered from an extremely small sweet spot. Needless to say, that’s a lot of different formats.

The LFE Channel

Before we look at the different immersive speaker formats in detail we must look at the concept of the LFE, or “Low Frequency Effects” channel. All immersive playback formats designated as “x.1” (such as 5.1 or 7.1) offer this dedicated subwoofer channel (the “.1” is the subwoofer). In film production, this is sometimes referred to as the “Boom” channel because that’s what it’s there for—to enhance the low frequencies of a film so you get the extra seat-shaking effect from an earthquake, plane crash, explosion, or any chest-thumping moment on screen that requires these low frequencies. (Many people believe that the richness and power of modern motion picture sound systems is what’s keeping the theatrical exhibition industry alive. I would agree.)

The LFE channel, which has a frequency response from about 30Hz to 120Hz, is unique in that it has an additional 10dB of headroom built into it. This is needed to accommodate the extra power required to reproduce the low- frequency content without distortion.

Bass Management

Subwoofers in these systems also include what’s known as a bass manager circuit (sometimes called bass redirection) that takes all the frequencies below about 80Hz (the frequency varies) from the main channels and the signal from the LFE channel and mixes them together into the subwoofer (see Figure 7.1).

The reason for this is, since there’s already a subwoofer available in the playback system, you might as well take full advantage of it for more than just the occasional low-frequency effect. This enables the effective response of the entire playback system to be lowered to about 30Hz, assuming the subwoofer is large enough. This approach is not necessary in

a Dolby Atmos system however, since all channels can be full range and may not require the bass extension from the subwoofer.

Since the overwhelming majority of consumer surround systems (especially the inexpensive ones) contain a bass-management circuit, if you don't mix with one in your monitor path you may not be hearing things the way the people at home are. And, since the bass manager provides a low-frequency extension below that of the vast majority of studio monitors, people at home may actually be hearing things (such as unwanted rumble) that you can't hear while mixing.

Diagram Description automatically generated

Figure 7.1: The LFE channel.

TIP: In the various configurations, the first number designates the number of main channels. If a low-frequency effects channel is used, the channel designation becomes "x.1" (or 5.1, for example); if a low-frequency effects channel isn't included in the format, it becomes "x.0" (or 4.0 for example).

1st Generation Immersive Audio Formats

Before we can discuss the panning technique for immersive audio, it’s important to be familiar with the various formats that are available. Immersive audio can be broken down into three distinct eras, with the first two generations being more “surround” in nature rather than fully immersive. This is because, for the most part, they’re based on audio coming directly from the available speaker channels (you’ll see the differences soon).

The first generation was the “x.0” formats, or the ones not using a subwoofer. All the formats in this generation were also delivered in analog.

Three-Channel (3.0)

This is an early Dolby Surround encoding format utilizing stereo front speakers and a mono speaker in the rear for surround (see Figure 7.2). It was first introduced in 1982 as a way to encode surround information onto CDs and VHS videotapes.

A picture containing diagram Description automatically generated

Figure 7.2: Dolby Surround

LCRS Four-Channel (4.0)

LCRS stands for Left, Center, Right, Surround and is the basic cinema setup of three speakers behind the screen in front of the audience fed by separate channels, and a single mono surround channel that feeds multiple speakers throughout the theater (see Figure 7.3).

A picture containing text, electronics, keyboard Description automatically generated

Figure 7.3: LCRS four-channel surround.

Quadraphonic Four-Channel (4.0)

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Figure 7.4: Four-channel quad surround.

The first true consumer surround format called “Quadraphonic” (or just “Quad” for short) was introduced in the early 1970s. It basically consisted of a stereo pair in front of the listener and another stereo pair behind the listener. The format never caught on primarily because there were two

competing delivery methods, which caused consumers to hesitate purchasing one over the other for fear they'd pick the standard that that was less popular and therefore wouldn't be supported (see Figure 7.4).

Five-Channel (5.0)

Dolby developed its Pro Logic encoding especially for delivery of multichannel audio to the home, and gradually the number of channels it could encode evolved from three-channel (stereo with a single surround), to four-channel (LCRS with a single surround), and finally to five-channel (LCRS with stereo surrounds). This became a popular format for television audio, especially when it became possible to deliver it digitally (see Figure 7.5).

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Figure 7.5: 5.0 surround.

2nd Generation Immersive Audio Formats

The second generation of surround technology were the "x.1" formats which incorporated a subwoofer into each system. These were also the first to be delivered in a digital form, such as via a DVD or Blu-Ray disc or over

digital electronic transmission for television. Second generation formats also experimented with more channels in an attempt to achieve a higher degree of realism.

With this generation, the limiting factors weren’t the quality of the product or the capabilities of the format. The main problem here was in the execution, specifically on the consumer side, where the speaker systems were rarely positioned or calibrated properly in the home so the effect was disappointing as a result.

Six Channel 5.1

The six-channel 5.1 surround was the gold standard for many years in both theater sound and home theater. The format consists of three channels across the front (L,C,R), two surround channels (left surround or Ls, and right surround or Rs), and a low-frequency effects channel (LFE). (see Figure 7.6).

A picture containing diagram Description automatically generated

Figure 7.6: 5.1 surround.

The 5.1 format dates back to 1976, but has become the dominant standard for surround sound since the arrival of the DVD in the early 1990s. Although many other formats were introduced (as you're about to see), they're all derived from the basic 5.1 setup. It's only now that Dolby Atmos is beginning to surpass the format in use and awareness.

Seven Channel 6.1

Even though 5.1 remained the standard surround format for theatrical motion pictures for some time, many film mixers complained that the 5.1 format was too limiting — they couldn't easily localize effects in the house with only two surround channels. In response to this, the 6.1 format was created to offer a center rear surround channel in an effort to improve the localization (see Figure 7.7).

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Figure 7.7: 6.1 surround.

Eight Channel 7.1

As both listeners' and mixers' tastes became more sophisticated, it was determined that the stereo rear channels of 5.1 didn't provide enough of a

sonic immersion to satisfy moviegoers or home-theater lovers. The eight-channel 7.1 attempts to change that limitation with the inclusion of additional surround channels on the sides (see Figure 7.8).

Shape Description automatically generated with medium confidence

Figure 7.8: 7.1 surround.

Eight Channel 7.1 SDDS

For a brief time, Sony offered their own eight-channel format known as “Sony Dynamic Digital Sound,” or SDDS. This was a 7.1 format that was configured somewhat contrary to what you might expect. Instead of additional surround channels, five channels were used across the front. One of the advantages to this format is that it provided the higher sound pressure level needed for larger theaters, but many also argued that it provided increased panning precision across the front speakers as well (see Figure 7.9).

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Figure 7.9: 7.1 SDDS surround

Twelve Channel 10.2

While he was the chief scientist at Lucasfilm, Tomlinson Holman originally coined the term “5.1” In the following years, he set out on his own, founding his TMH Labs, where he took motion picture sound engineering much further than the rest of the industry.

While most firms were still experimenting with 5.1, TMH Labs leaped far ahead of the pack when they created the 10.2 format, which is the same as the 7.1 format except for the addition of stereo LFE channels, a center rear channel, and stereo height channels placed in the front over the screen (see Figure 7.10). This was a direct precursor to the immersive audio setups that we use today.

As an aside, Tom is also responsible for the THX sound standard that you see in movie theaters. THX stands for “Tom Holman eXperiment.”

Shape Description automatically generated

Figure 7.10: 10.2 surround

Twelve Channel 11.1

In an attempt to totally immerse the listener in sound, the 12-channel 11.1 system was developed. This format adds additional side surround channels as well as stereo height channels above the screen (see Figure 7.11).

Shape Description automatically generated

Figure 7.11: 11.1 surround.

All of these formats are based on sound coming out of speaker channels, so panning is always measured on a single horizontal plane. Even with height channels, there was little ability to pan in 3 dimensional space given the production tools that were available, which was one of the first challenges that 3rd generation immersive formats tackled.

3rd Generation Immersive Audio Formats

Surround sound truly becomes immersive audio with the introduction of the multichannel Dolby Atmos system in 2012. Thanks to a whole new set of digital production tools and true overhead height speaker channels, mixers are finally able to provide the audio experience that the earlier generations weren’t able to approach. Today the capabilities are so vast that, truly, the only limitation is the mixer’s imagination.

Just as in the Quad days though, there are multiple technologies vying for the same consumer with the addition of Sony 360 Reality Audio, DTS-X, and L-Acoustics L-ISA formats coming on the market (although L-ISA is used mostly for concert and theater sound). So far, Dolby Atmos has a significant lead in installations, available tools, and song releases, so that’s what we’re going to concentrate on here.

You might be thinking, “Why is immersive audio so different from the previous surround formats?”

Probably the biggest difference in 3rd generation systems is that panning went from channel-based to object based. That means that a mixer is no longer panning towards a speaker but instead into a three dimensional space in the listening area. Thanks to more speakers being used (up to 64, which is seen mostly in theaters), these objects can envelop the listener in sound in a more seamless and natural way than ever before.

And unlike earlier immersive formats, integration into the typical home is now much easier and more attractive. Today, a simple soundbar, a subwoofer, and a couple of wireless speakers would not only get the seal of approval from the family member most worried about the decor, but it would also provide a very enjoyable listening experience for everyone in the room as well.

An Introduction To Dolby Atmos

A Dolby Atmos playback system takes listener immersion and mixer panning ability to a whole new creative level. That said, it also has a number of additional advantages beyond the creative ones, such as:

It’s both future-proof and backwards-compatible with previous 2nd generation formats.

It’s easy to monitor between a wide variety of immersive formats.

It can also be played back in regular stereo on non-Atmos-enabled playback devices.

Although the system features up to 128 channels routed to 64 speakers (which, as we mentioned, is most often utilized in theaters), the normal music production space utilizes a 7.1.4 (as seen in Figure 7.12) or 9.1.4 playback system. That means a 7.1 or 9.1 system with the addition of 4 height channels positioned in a square above the mixer. This speaker configuration coupled with the addition of the Dolby Production Suite panner allows the mixer to place an object anywhere in a 3D space in the room.

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Figure 7.12: A typical 7.1.4 immersive speaker setup

Beds And Objects

While mixing in Generation 1 and 2 formats consisted of bussing or panning sounds between speaker channels, Generation 3 immersive formats use a different concept called “object-based.” That means that you have the same traditional panning to channels available as before, but also have the ability to pan other mix elements in a total 3D space. These are what’s known as beds and objects.

Beds

Atmos (as well as the other immersive formats) still allows you to pan in a horizontal plane towards any speaker that you’d like, but also adds two additional overhead channels. This is known as a Bed. The Bed audio can be anything from stereo up to 7.1.2 (the .2 being the Left and Right Overhead channels).

While you can have multiple Beds if you desire, most mixers just use one that’s oriented more towards the front of the room. Some call this the “front wall” of the mix.

When it comes to music, Beds are particularly effective for low frequency information, since the only way to assign a mix element to the LFE channel is through a Bed. Mix elements that work well in a Bed include individual drum tracks, percussion instruments, bass, lead vocals, and reverb.

That said, some mixing approaches forgo the Bed entirely and only use Objects, and that’s what we’ll look at next.

Objects

Objects have the unique ability to pan anywhere in a 3D space in the room, rather than towards just the speakers. Objects can be panned in three coordinates - an X plane (left to right), a Y plane (front to back) and a Z plane (up to down), but even better, the size of that space can also be easily controlled with a simple Size parameter control (Figure 7.13).

Graphical user interface, application Description automatically generated

Figure 7.13: The panning and size parameters

Objects are generally used for more precise and dynamic placement of mix elements like synths, guitars, percussion and effects.

A huge piece of Object-based mixing is that all of the size and positional data is captured as metadata, which will be used in many ways later, as you'll soon see.

TIP: If you're mixing to multiple formats, it's best to begin in the most immersive format of Atmos (like 7.1.4) rather than in stereo.

Rendering

Where Atmos can get confusing is that it's actually a two part system consisting of the DAW that you're working on, and another essential processor called the Renderer (Dolby calls it the Renderer Mastering Unit or RMU). Instead of routing to a 2-channel or multichannel output bus in your DAW, you route up to 128 audio channels to the Renderer instead.

The Renderer is actually the place where all the various output formats can be created simultaneously, and where the audio is monitored from. This takes place during both content creation and consumer playback (see Figure 7.14).

Diagram Description automatically generated

Figure 7.14: The Dolby Atmos Renderer

During mixing the Renderer can be hosted on the same computer as the DAW software, but for large sessions, it’s more efficient to host it on a separate computer. When that happens, the DAW computer and the Renderer computer are connected with the Dolby Audio Bridge either via multichannel MADI or Dante connection that syncs the two together thanks to a 129th channel carrying time code. This becomes a lot more complex setup in that it requires additional hardware such as a MADI or Dante router, a time code generator and master clock (not to mention the multichannel monitor controller needed to adjust the volume of all the speakers that you need anyway).

Regardless of how the Renderer is hosted, one of its main functions is to allow you to listen to the mix in any number of formats without having to patch or reconfigure anything. That means you can hear it in 7.1.4, or 5.1, 3.0, or stereo by just selecting the appropriate output.

Logic Pro, Nuendo, and DaVinci Resolve actually have the renderer built into the DAW but the feature set is not as complete as with the full Dolby Renderer, which must be purchased separately (see Figure 7.15).

Graphical user interface Description automatically generated

Figure 7.15: The Dolby Atmos Monitor Application

Courtesy Dolby Laboratories

Unlike with Generation 1 and 2 playback systems, Generation 3 consumer playback systems are intelligent, and combined with the Renderer, provides a superior immersive experience even with few speakers available. The Renderer that's built into in a smart consumer device will sense the speaker configuration available, then derive a format from the Atmos file that best reproduces the mix in that environment. Devices like the Apple TV4k, Roku, Amazon Firestick, and Amazon Echo Studio support Atmos, and services like Apple TV+, Netflix, Amazon Prime, and HBOMax currently stream the format.

Binaural Playback

Setting up a full 7.1.4 system can be daunting not only from a cost perspective — if you have to purchase all the speakers and monitor controller — but from a studio space perspective as well. Luckily there's a way that you can monitor fairly successfully with just a set of good headphones by using the binaural settings on the renderer.

The Binaural Renderer renders all of the Bed and Object audio to create a compelling immersive mix over headphones using head-related transfer function (HRTF) filters. This replicates the experience of listening to immersive audio on speakers as closely as possible while wearing headphones. A mix created while monitoring via loudspeakers will also reliably translate to the binaural renderer.

Unlike the Renderer output meant for speakers, the Binaural Render has three mode settings that lets you determine the distance at which each object and the beds are perceived. These settings are OFF, NEAR, MID and FAR and there's a big difference between them.

TIP: It's best to monitor the Dolby renderer's binaural mix with all distance settings set to MID first before finishing your binaural mix.

Considering that most listeners will be hearing your immersive mix using headphones or earbuds, monitoring the binaural mix is important. That said, it's actually possible to mix binaurally and have it translate well when using speakers later.

Obviously it's always best to check with both speakers and headphones, but many mixers have reported good results using only headphones, especially when they were just getting started in Atmos mixing.

The Apple Music Conundrum

Atmos mixes are now available on multiple streaming services like Amazon Music, Tidal and Apple Music. Amazon and Tidal currently utilize the Dolby Atmos codec as you hear it from the Renderer so your mix will sound true to what you heard while mixing. Apple Music is a different story however.

Although Apple Music also uses the Dolby Atmos codec, the platform uses it own Spatial Audio Visualizer that's different from Dolby's. While providing an immersive mix, this can present a substantially different representation of the mix. Many artists and mixers are surprised when they hear what their mix sounds like on the platform, so it pays to be aware of the differences before a mix is sent to the service.

The reason why mixes sound different on the platform is that Apple's Spatial Audio decoding algorithm ignores the binaural distance settings of the Dolby Atmos encoder.

TIP: Be aware that Apple Music may play a Spatial Audio mix about 10dB louder than other platforms, making it more likely to trigger the playback limiter and change the sound as a result. You can stop this from happening by setting your iOS device to Soundcheck ON for the best experience.

Mixing Considerations

We're at the beginning of what many consider the age of true immersive audio, and while many of the tools that we're used to having while mixing in stereo are available, only a few plugins can provide the full multichannel experience. That said, immersive audio requires less of what's normally required in a stereo mix. For instance, because the spread in the spatial environment is so wide, much less EQ and compression is needed, since mix elements no longer have to be squeezed into a relatively small stereo space. (Remember our cowboys?)

The Loudness Limit

But there are other considerations as well. Atmos mixes have a fixed target loudness of -18LUFS. While this might seem considerably lower than modern stereo mixes, there's a good reason for this setting. Since Atmos is great at folding down a 7.1.4 mix to fewer channels (even down to stereo), that also means the summing of the 12 or 14 immersive channels down to 2 means the level will automatically increase. Staying at -18LUFS or below provides enough headroom so you'll never have to worry about an overload occurring after a fold down.

Because Dolby Atmos allows for the full frequency range in all speakers, it's important to be aware of where bass-heavy Objects are panned. Excessive low-end in an Atmos mix may take up too much headroom and cause a mix to be rejected as a result.

Atmos Files

There are several types of files associated with an Atmos mix. Before we discuss them in detail, be aware that the resolution for a Dolby Atmos mix can be up to 96kHz/24 bit and that's what's encoded into the Atmos session file. The sample rate of the tracks in your DAW can be different (like 96kHz for instance), but eventually the sample rate will be transcoded to 48kHz for streaming distribution.

The file that your DAW uses during production is a .atmos ("dot-atmos") master file. In other words, it will be named "xxxxx.atmos." This appears on your computer as a folder with usually three or four files inside containing the audio and the metadata.

When it’s time to deliver your mix to a streaming distributor, you export your project as an Audio Definition Model Broadcast Wave Format, or ADM BWF, file. ADM BWF files contain a multichannel audio file containing the surround bed and the individual mono and split stereo object tracks, as well as all the metadata required for reproducing the mix on systems or devices compatible with Dolby Atmos.

A smaller MP4 file can also be generated that can be used on Apple devices for checking the mix using Apple’s Spatial Audio codec, or consumer playback equipment, like a Blu-Ray player or Soundbar.

While it’s true that a stereo mix can be derived from your Atmos file, you still need to produce a separate stereo master for Apple Music as well. The reason is that your stereo master runs in parallel to your Spatial Audio as as it’s being played. This enables you to switch on-the-fly between the two formats, but it also requires that the two be as tightly synchronized as possible.

The Renderer also allows you to bounce your Atmos mix to other discrete surround audio file formats as well if required.

Content Creation Tools

Dolby provides two packages that include the rendering software, monitoring software, and a dedicated Atmos panner. These include the Dolby Atmos Production Suite and the Dolby Atmos Mastering Suite.

The difference is that the Production Suite is intended for the DAW and Renderer to use on a single computer, while the Mastering Suite is intended for larger installations with multiple licenses and different workstations.

The Production Suite costs $299 with a 90 day free trial, while the Mastering Suite is $999. Both can be purchased at dolby.com/music/create.

While an Atmos panner is part of either suites, several DAWs already feature a native Dolby Atmos panner, including:

Apple Logic Pro

Avid Pro Tools Ultimate

Blackmagic Designs Resolve

Steinberg Nuendo

Merging Technologies Pyramix Premium

Besides that, several DAWs also natively feature the panner as a plugin. These include:

Ableton Live

Apple Logic Pro

Avid Pro Tools (both Ultimate and non-Ultimate)

We’ve just dipped a small toe in the Atmos waters as it’s a deep subject and could easily fill a book on its own. The good thing is that with a DAW like Logic Pro X and a set of headphones, it’s a lot simpler and cheaper to get into than you might’ve thought.

Immersive Audio Mixing Approaches

Regardless of the speaker configuration, mixing for any type of immersive audio is different from plain old stereo. In may ways, stereo is a lot more work as we’re forced to use EQ and compression in order to fit all the mix elements into a tight spatial window (it’s even worse for mono).

Immersive audio provides the mixer with so much more freedom because of all the spatial room available thanks to the extra speakers. Because of all

the new space, less signal processing is needed, which actually takes some experienced mixers time to get used to.

Let’s look at the different approaches and considerations that having more speakers to mix to can present.

Differences between Immersive Audio For Picture And For Music

Regardless of the speaker format that you’re working in, there’s a difference between mixing for picture and mixing for music-only. Normally in the theater, all of the primary sound information comes from the front speakers, and the surround speakers are utilized mostly for ambience info in order to keep your attention — your eyes — on the screen. The LFE is intended to be used primarily for special effects such as explosions and earthquakes and is therefore used infrequently.

One of the reasons that the surround speakers don’t contain more source information when used with picture is a phenomenon known as the “exit-sign effect.” This means that your attention is drawn away from the front screen to the exit sign over the emergency exit at the side of the theater when the information from the surrounds is too loud or the front panning is too wide.

Music-only immersive audio has no screen to focus on and therefore no exit-sign effect to worry about. Take away the screen, and it’s now possible to utilize the surround speakers for more creative purposes.

Immersive Music Mixing Panning Approaches

There are two schools of thought about how immersive audio for music should be mixed, regardless of the speaker format. The “audience” approach puts the music mostly in the front speakers and the venue ambience or simulated ambience in the surround speakers, just as if you were sitting in a club or concert hall. This method may not utilize the LFE channel at all and is meant to reproduce an audience perspective of the musical experience from what could arbitrarily be called “the best seat in the house.”

The second method is the “middle of the band” approach. In this case the mix elements are spread all over the room via the front, back and overhead channels or front wall, and that puts the listener in the center of the band and envelopes him or her with sound. This approach usually results in a much more dramatic soundstage that is far bigger-sounding than the stereo that we’re used to.

What Do I Put In The Center Channel?

In film mixing, the center channel is used primarily for dialog so the viewer/listener doesn’t get distracted by movement in the soundfield. In music, however, its use prompts debate among mixers.

No Center Channel

Many veteran engineers who have mixed in stereo all their lives have trouble breaking the stereo paradigm to make use of the center channel. These mixers continue to use a phantom center from the left and right front speakers, or prefer not use it at all (see Figure 7.16).

A picture containing text, clock, watch Description automatically generated

Figure 7.16: No center channel

Isolated Elements In The Center Channel

Some mixers prefer to use the center channel to isolate certain elements, such as lead vocals, solos, and instruments with high-frequency information that infrequently appears. While this might work in some cases, many times the isolated elements seem disconnected from the rest of the soundscape unless they're at least partially bled into the other channels (see Figure 7.17).

Diagram Description automatically generated

Figure 7.17: Isolated center channel

The Center As Part Of The Whole

Mixers who use the center channel to its fullest find that it can act to anchor the sound and eliminate any drifting phantom images. In this case, all front and rear speakers have equal importance, with the balance changing the sound elements placed in the soundscape (see Figure 7.18).

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Figure 7.18: Center as part of the whole

"I guess if I were to encapsulate the rule, the things that I used to put in the middle I put everywhere now. Bass, kick drum, snare drum, lead vocal—all the stuff that has a lot of mono correlated information goes a bit to every speaker, except maybe the center. If I put something in the front, I will very rarely put it in the center and the left and the right. I will put it in the center and the surrounds if I want to pull it more into the middle of the room. If I want something off to the side of the room, I'll go left, right, and right surround so it leans to that side."

—Nathaniel Kunkel

“I like as much information coming out of the surrounds as I do the front, so I’m still as aggressive as I was when I first started, maybe even more so. The only thing that’s really changed for me is how I use the center speaker. I try to use it a little more than I have in the past. I’ve found that when listening in a car, the center speaker is a little more important than it is in the home.”

—Elliot Scheiner

“So I thought, Why don’t I put the vocal in the center monitor most of the time, and the only other things that enter into that monitor are double vocals or harmonies or maybe even a solo instrument? Then I’ll bleed out a little bit of the center vocal into the left and right fronts so if Uncle Bob comes over to the house and sits at that end of the couch, he’s not missing the lead vocal. Then I’ll use divergence and spill a little of that lead vocal into the rear monitors also for that purpose.”

—Greg Penny

What Do I Send To The LFE Channel?

Anything that requires extra low-frequency bass extension can be put into the subwoofer via the LFE channel. Many music mixers put a little kick and/or bass there if it’s used at all.

That said, it might be better not to even use the LFE channel unless you’re positive that the subwoofer is calibrated correctly. An uncalibrated subwoofer can cause big surprises in the low end when the track is later

played back on the typical home-theater setup. If you don't use the sub when mixing, the low frequencies under 80Hz are naturally folded into the playback subwoofer, resulting in a smooth and even response.

TIP: The frequency response of the LFE channel only goes up to 120Hz, so you may have to put any instrument that's sent into the LFE into the main channels as well to gain some definition.

"I run the speakers full range so I don't have to put all that much in the LFE. They [consumer electronics manufacturers] all have their own ideas about what bass management is, so I just ignore it and use the LFE to put a minimum of stuff in. Generally, I like to get things to sound how they should sound on my monitors without the LFE."

—Elliot Scheiner

"The LFE is a low-frequency effects track. It's used when you have run out of low-frequency headroom in your other channels, so I only go to the .1 when I cannot put any more level on the main channels and I want more bass."

—Nathaniel Kunkel

With the current 7.1.4 formats, how we approach panning has completely changed. While there is more to consider during an immersive mix than ever before, the complexity that's currently involved in achieving exceptional results should reduce over time. As we gain more collective knowledge and experience, and consumers continue to embrace the format, it will be no time before we approach immersive mixing with the same degree of comfort and confidence that we have for stereo mixing today.

8

The Dynamics Element: Compression, Limiting, Gating, And De-Essing

At one point in the history or recorded music, the control of the volume envelope of a sound, otherwise known as its dynamics, would not have been included as a necessary consideration of a great mix. In fact, dynamics control is still not a major concern among many professionals in the orchestral and jazz mixing world. Today's modern mixes have different demands however, so the manipulation of dynamics plays a major role in the sound in most contemporary music, if for no other reason than almost nothing can affect your mix as much and in so many ways as compression.

"I think that the sound of modern records today is compression. Audio purists talk about how crunchy compression and EQ are, but if you listen to one of those jazz or blues records that are done by the audiophile labels, there's no way they could ever compete on modern radio even though they sound amazing. Every time I try to be a purist and go, 'You know, I'm not gonna compress that,' the band comes in and goes, 'Why isn't that compressed?'"

—Jerry Finn

Types Of Dynamics Control

An audio source's dynamic range is controlled by the skillful use of four essential tools in the mixer's arsenal:

Compression

Limiting

Downward expansion (gating)

Upward expansion (transient design)

Just to clarify what each one does, here’s a brief description of each.

Compression

What we know as compression is more properly called dynamic range compression because it’s the process of taking an audio source signal with a large dynamic range (meaning the softest to loudest points in the signal) and making it smaller. This is done by lowering the loudest portions of the program to more closely match the lowest ones so the volume level is more constant.

Compressors work on the principle of gain ratio, which is measured on the basis of input level to output level, and is set by using the Ratio control (see Figure 8.1).

A picture containing text, black, meter, device Description automatically generated

Figure 8.1: Typical compressor parameters

Courtesy Apple Inc.

As an example, a ratio of four-to-one (or 4:1 as you'll see it on the compressor), means that for every 4dB that goes into the compressor, 1dB will come out (see Figure 8.2). If the ratio is set at 8:1, then for every 8dB that goes into the unit, only 1dB will be seen at the output.

A ratio of 1:1 results in no compression at all.

Graphical user interface, application Description automatically generated

Figure 8.2: Compressor ratio

The Threshold control determines the point in the signal level where the unit will begin to compress (see Figure 8.1). As a result, threshold and ratio are interrelated, and one will affect the way the other works. Some compressors (such as the Universal Audio LA-3A in Figure 8.3) have a fixed ratio, but on most units the parameter is variable.

A picture containing text, device, meter, control panel Description automatically generated

Figure 8.3: A Universal Audio LA-3A showing 6dB of compression

Courtesy Universal Audio

Most compressors have Attack and Release parameter controls (see Figure 8.1) that determine how fast or slow the compressor reacts to the beginning of the signal envelope (the attack) and end of the signal envelope (the release). These controls are especially important because each setting is crucial to how the compression works and sounds. Because the settings on the attack and release controls can be tricky if you’re not sure how to use them, some compressors have an Auto mode, which will set the attack and release parameters according to the dynamics of the input signal. Although Auto mode can work relatively well, it doesn’t allow for the precise settings that may be required to properly control certain source material. Many mixers will use the Auto mode as a starting place and fine-tune their settings from there.

While most compressors have some control over the volume envelope, some compressors have a fixed attack and release (such as the dbx 160 series—see Figure 8.4), which gives it a particular sound as a result.

A picture containing text, device, meter, control panel Description automatically generated

Figure 8.4: A dbx 160 plugin

Courtesy Universal Audio - dbx

Many compressors also have a Knee parameter, which sets how fast the compressor will begin to compress after the signal reaches the set threshold (see Figure 8.5). A low value (typically, measured in dB, so 0dB would be the lowest) means that the compression will begin instantly at threshold. A higher setting will gradually ease into the compression, which may sound better on some types of program material, like vocals, for instance, where the compression is meant to be less apparent.

Picture 72

Figure 8.5: The compressor knee parameter

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When a compressor operates, it actually decreases the gain of the signal, so there’s another control that allows the signal to be boosted back up to its original level or beyond called Make-Up Gain or Output (see Figure 8.1).

All compressors have a gain-reduction meter to show how much compression is occurring at any given moment (see Figure 8.1). This can look like a normal VU or peak meter, but it reads backwards. A compressor meter reading 0dB means that the signal is below threshold so no

compression is taking place and any movement to the left or down into the minus range shows the amount of compression that's occurring. Stated another way, a meter that reads -6dB indicates that there is 6dB of compression taking place at that instant (see Figure 8.3).

Many compressors also have a feature known as a Sidechain (sometimes called a "Key" input), which is a separate input into the compressor so that other signal processors can be connected to it (see Figure 8.6). This sidechain can have a number of useful purposes such as when an EQ is connected to make a makeshift de-esser. If only the frequencies in the upper range are boosted, the loud "S" sounds from a vocalist will be attenuated when they exceed the compressor's threshold. We will discuss "De-essing" in greater detail later in this chapter.

Graphical user interface Description automatically generated

Figure 8.6: The sidechain input and filters

You can also connect a delay, reverb, or any other signal processor to the sidechain to create some unusual, level–dependent effects. Because a sidechain isn't needed for most everyday compressor operations, many manufacturers elect not to include sidechain connections on hardware units but may include one on the plugin version.

A sidechain can also be used to "duck" or attenuate another instrument when a new one enters. For instance, if you connected a send of a vocal track into

the sidechain of a compressor inserted on a loop track, the loop would lower in volume whenever the vocal entered and then return to its normal level when the vocal stopped (see Figure 8.7). This is what happens at an airport when the music is automatically lowered for a gate announcement and then returns to its normal level when the announcement is finished.

Diagram Description automatically generated

Figure 8.7: Connecting to the compressor sidechain to duck a track

Compressor Differences

There are a number of different types of compressors that every engineer should be familiar with but there’s not a lot of information available about the differences between these devices and what each is good at. Most compressor plug-ins today are based on the four different electronic building blocks that were used to build a compressor back in the days of analog hardware. Here’s a brief look at their characteristics and what we can do with them.

Optical. A actual tiny light bulb and a photocell were used as the main components of the compression circuit. The time lag between the bulb and the photocell gave it a distinctive slow attack and release time (like in an LA-2A-style compressor) that made it useful when large transients aren’t present (like vocals). It’s transparent, tightens up a track without being noticed, and adds warmth. Limitations: An optical compressor doesn’t

control fast transients very well, and tends to pump (compression out of synch with the tempo of the song) with signal content that has a lot of low end frequencies.

FET. A Field Effect Transistor was used to vary the signal gain, which had a much quicker response than the optical circuit. (A Universal Audio 1176 is a good example.) It has very fast attack and release with lots of control thanks to the additional Attack, Release and Ratio parameter controls. It also tends to be very agressive sounding, and works best for punch and snap, but it also adds the most color, although this color can can be very warm and rich. Typically an FET-style compressor won't work well on the mix buss. Limitation: Not very transparent.

VCA. A Voltage Controlled Amplifier circuit was a product of the 1980s that had both excellent response time and more control over the various compression parameters. Aggressive, with extreme settings and extended parameter functions, a VCA works well on material that has a lot of peaks in the program. It also works well in situations where the transients are already controlled (like the mix buss), and it's excellent for adding punch to drums and bass. Limitations: Doesn't always smooth out volume as needed. Can be thin sounding.

Vari-Gain. The Vari-Gain compressors are sort of a catch-all category because there are other ways to achieve compression besides the first three (such as the Fairchild 670 and Manley Vari-Mu). This style of compressor takes time to react, which adds a "glue" to the mix (like the mix buss or subgroup) and adds warmth and fatness. One major characteristic of a Vari-Gain device is that the compressor ratio increases with gain reduction, so the louder the transient, the harder it's compressed. Limitations: Slow attack and release, and won't solve dynamic issues or increase the punch of a sound.

Today, we can add one more type of compressor to our toolbox:

The Non-emulation Digital Compressor. While it’s great to create a compressor plugin based on a previous hardware model, today’s digital compressors can do so much more. Almost every developer now has a compressor that goes way beyond what a hardware emulation can do and that results in parameters and sound that never could have been achieved using analog hardware. There is a limitation though. While some digital compressors can provide hardware emulation as well, not many are particularly good at it.

As you would expect, each of the above has a different sound and different compression characteristics, which is the reason why the settings that worked well on one compressor type won’t necessarily translate to another. The good thing about living in a digital world is that all of these various hardware compressors have been replicated by software plugins, so it’s a lot easier (not to mention cheaper) to make an instant comparison on a track and decide which works better in each particular situation.

Multi-Band Compression

Multi-band compression splits the input audio signal into two to six separate frequency bands, each with its own compressor (see Figure 8.8). The main advantage of a multi-band is that a loud event in one frequency band won’t cause any gain reduction in the other bands. That means something like a loud kick drum will cause the low frequencies to be compressed, but the mid and high frequencies are not affected. This allows you to get a more controlled, hotter signal with far less compression than with a typical single-band compressor.

With the rise of dynamic EQ plugins, multiband compressor use is falling by the wayside since it is not as versatile.

A picture containing text, metal, scoreboard Description automatically generated

Figure 8.8: A multiband compressor

© Courtesy Universal Audio

Limiting

Compression and limiting are closely related, the main differences being the setting of the ratio parameter and the application. Any time the compression ratio is set to 10:1 or greater, the result is considered limiting, although most true limiters have a very fast attack time as well.

A limiter is essentially a brick wall in terms of level, allowing the signal to get only to a certain point and not beyond. Think of it doing the same thing as a speed governor used on 18-wheel trucks owned by a trucking company to make sure they're not driven beyond the speed limit. Once you hit 65 mph (or whatever the speed limit in a particular state), no matter how much more you depress the gas pedal, the truck won't go any faster. The same logic applies to a limiter: once you hit the predetermined level (set by a Ceiling control), no matter how much you try to push beyond it, the level pretty much stays the same.

Most modern digital limiters (either hardware or software) have an internal function known as "look ahead" that allows the signal detection circuitry to look at the signal a millisecond or two before it hits the limiter detection path (Figure 8.9). This means that the limiter acts extremely fast and just about eliminates any overshoot of the predetermined level, which can be a problem with analog limiters because they react much more slowly to transients.

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Figure 8.9: Universal Audio Precision Limiter

Courtesy Universal Audio

Limiting is used a lot in sound reinforcement for speaker protection and on many powered studio monitors as well to prevent blown speakers from extremely high playback levels. During mixing, many engineers who feel that the bass guitar is the anchor for the song want the bass to have as little dynamic range as possible. In this case, limiting the bass by 3dB or so (depending on the song) with a ratio of 10:1, 20:1, or even higher can achieve that result.

De-Essing

One of the major problems when mixing vocals is a predominance of S's that comes from a combination of the mic being too close to the vocalist and the type of mic being used (usually a condenser that has a presence peak in the sibilance range), the EQ added during mixing to make it cut through the mix, and the use of heavy compression. Sometimes this is not an issue until it's time to mix, when a compressor is put on the vocal to even out the level and, all of a sudden, every "S" from the singer seems ear-piercing. This effect is what's known as "sibilance" and is totally undesirable.

The way to combat sibilance is to use a de-esser, which is a unit that compresses just the vocal S frequencies that usually fall somewhere between 2kHz and 10kHz, depending upon the situation.

A de-esser can be created using a compressor with an equalizer plugged into the sidechain as stated above, but it's usually a dedicated unit designed just for this purpose. While many hardware de-essers are limited to two parameter controls, Threshold and Frequency, software de-essers are much more sophisticated, allowing for pinpoint control that a hardware unit just can't duplicate. (Figure 8.10) The Threshold control is similar to the control found on a compressor/limiter in that it sets the level when the de-essing process begins. The Frequency control allows you to fine tune the frequency to exactly where the S's occur on a particular recording.

Graphical user interface Description automatically generated

Figure 8.10: An SPL De-esser plugin

Courtesy Plugin Alliance-SPL

Gating

Although not used very often in the recording studio now that so much processing, editing, and sound replacement is possible in digital workstations, gates are still used in certain studio situations and are very common in sound reinforcement (see Figure 8.11). More correctly called a downward expander, a gate keeps a signal turned off or attenuated at a lower level until the signal reaches a predetermined threshold, at which time it opens and lets the full sound through.

The gate can be set to mute the sound completely when it drops below a threshold or to just lower the level to a predetermined amount which is determined by the Range control. Depending on the situation, just turning the level down a bit sometimes sounds more natural than turning it off completely, although the total silence can sometimes be used as a great effect.

Graphical user interface Description automatically generated

Figure 8.11: A Pro Tools Digirack gate

Courtesy Avid

For a long time, engineers would use a gate (sometimes called a noise gate) to reduce or eliminate problems on a track such as random noises, lip smack, buzzes, coughs, or other low-level noises off-mic. On loud electric guitar tracks, for instance, a gate could be used to effectively eliminate amplifier

noise [HUM] during times when the guitar player is not playing. Today, it's more precise to just clean up any noise on the tracks with editing.

Because most of these situations can now more effectively be dealt with in the DAW (see Chapter 12, "Advanced Techniques"), gates aren't used as much in the studio, although there's still a use for them in many live-performance situations.

Gates are still frequently used on drums to control leakage from tom mics in a mix, or to tighten up the sound of a floppy kick drum by decreasing the ring after it's struck by the beater.

A gate can also have a sidechain, an additional input called a "Key" or "Trigger" input which allows the gate to open when triggered from another DAW channel or processor. This can be very useful in a number of situations, as seen in the "Gating Techniques" section at the end of this chapter.

Just like compressors, modern hardware gates are very fast, but plugins have the advantage that they can be designed so that the sidechain senses the signal a millisecond or two before it arrives at the gate's main input, allowing it to begin opening just before the transient arrives. This look-ahead facility is only an advantage when dealing with sounds that have a very fast attack, so it's sometimes switchable or variable when it is provided.

Transient Shapers

The opposite of a downward expander is an upward expander, sometimes known as a transient designer or shaper. An upward expander takes any signal above the threshold and increases its gain, acting like the opposite of compression. Unlike compressors, transient shapers can transparently shape the attack and sustain characteristics of sounds.

With a normal compressor set to a 2:1 ratio, for every 2dB above threshold the output would be 1dB. With an upward expander, that ratio is reversed to 1:2, so for every 1dB above threshold the output will rise by 2dB. This increases the dynamic range by making loud parts or peaks louder, and has the effect of putting the transients back into a part that's been too compressed. This is often used on drum overheads, room mics and snare drums to emphasis the crack of the drum and lengthen its effect by increasing the Release control.

Some excellent transient shaper plugins include the Stillwell Audio Transient Monster and the one that started it all: the SPL Transient Designer (see Figure 8.12).

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Figure 8.12: An SPL Transient Designer plugin

Courtesy SPL

Clippers

A clipper (sometimes called soft clipper) is an interesting processor in that it carefully reduces momentary signal peaks in a way that makes the signal seem louder without actually adding any gain or otherwise changing the sound. It does this by gently reshaping the waveform instead of simply chopping off its peaks. If used moderately, it delivers acleaner result with fewer audio side effects than a typical peak limiter — which is why it's often used on drum tracks — but can work equally well on other mix elements or even across the entire mix as well. Just as with many other types of plugins, a little Clipping goes a long way.

Some clipper plugins to check out include the JST Clip and IK Multimedia's Classic Clipper.

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Figure 8.13: T-RackS Classic Clipper plugin

Courtesy IK Multimedia

Leveler

A leveler automatically detects level differences in a mix element and corrects them in real-time so you don't have to make manual or even automated gain adjustments. It's typically set to a "target volume level" and then it will increase or decrease the level over the length of the recording

while keeping it close to the set target level. The controls are fairly limited but that also means the setup is simple. Examples of well-regarded levelers include the Accusonics Voice Leveler, Waves Vocal Rider, and the built-in leveler in the Oxford Drum Gate.

Using Compression

If there is one recognizable difference between the sound of a demo or semi-pro recording and the sound of a finished professional mix, it’s the use of compression. As a matter of fact, the difference between the sound of one engineer’s mix and another’s often just a matter of how he or she uses the available compressors.

There are two reasons to add compression to a track or mix: to control the dynamics or as an effect.

Controlling Dynamics

Controlling dynamics means keeping the level of the sound even; in other words, lowering the level of the loudest passages so that there’s not too much of a difference between loudest and quietest.

Here are a couple of instances where this might be useful:

On a bass guitar. Most basses inherently have certain notes that are louder than others and some that are softer than others depending upon where on the neck they're played. Compression can even out these differences.

On a lead vocal. Most singers can't sing every word or line at the same level, so some words may get buried as a result. Compression can help every word to be heard.

On a kick or snare drum. Sometimes the drummer doesn't hit every beat with the same intensity. Compression can make all hits sound somewhat the same in level.

TIP: When controlling dynamics, usually a very small amount of compression (2 to 4dB) is used to limit the peaks of the signal.

"I like to compress everything just to keep it smooth and controlled, not to get rid of the dynamics. Usually I use around a 4:1 ratio on pretty much everything I do. Sometimes on guitars I go to 8:1. The kick and the snare I try not to hit too hard because the snare really darkens up. It's more for control, to keep it consistent. On the bass, I hit that a little harder, just to push it up front a little more. Everything else is for control more than sticking it right up in your face."

—Benny Faccone

Compression As An Effect

Compression can also radically change the sound of a track. A track compressed with the right compressor and with the correct settings can make it seem closer to the listener and bring more aggression and excitement. The volume envelope of a sound can be modified to have more or less attack, which can make it sound punchy, or to have a longer decay so it sounds fatter.

"Compression is the only way that you can truly modify a sound because whatever the most predominant frequency is, the more predominant that frequency will be the more you compress it. Suppose the predominant frequencies are 1k to 3k. Put a compressor on it, and the bottom end goes away, the top end disappears, and you're left with "Ehhhhh" [a nasal sound]."

—Andy Johns

"I usually spend the last two hours of my mix not doing much mixing but listening and then making little adjustments to the master buss compressor and hearing what the impact is to all the parts. Most of the effect is how it's putting the low- frequency information in check, which it does without the meter even moving at all. Even when you're hitting it very lightly, it still has a dramatic effect on the music."

—Bob Brockman

Placement In The Signal Chain

Where a compressor, limiter, de-esser, gate or expander is placed in the signal chain can have a dramatic effect on what that device does to the sound. If a compressor or limiter is placed after an EQ, every time the EQ is

changed it will affect the compressor settings. Plus, any frequency that’s boosted will have less of an effect because it will be turned down by the compressor. Likewise, a de-esser placed after an EQ may become ineffective if the midrange or high frequencies on the EQ are changed. Similarly, the threshold setting where a gate opens can change if the EQ is adjusted when it’s placed before it in the signal chain.

That’s why it’s best to place any compressor, limiter, or de-esser first in the signal chain, because the signal will receive all the benefits of the dynamics module and any other processor that comes after it (see Figure 8.9).

Diagram Description automatically generated

Figure 8.14: Dynamics placement in the signal chain

Setting The Compressor By Ear

In most modern music, compressors are used to make the sound “punchy” and in your face. The trick to getting the optimum punch out of a compressor is to let the attacks through and play with the release to elongate the sound. Fast attack times can sometimes reduce the transients, and therefore the high end of a signal, while very slow release times may make the compressor pump out of time with the music or even cause the signal to distort.

As it turns out, these parameters are often overlooked, as many engineers opt for either default settings or one that they tend to use in every situation. That might work in many cases, but this approach won't give you that little extra something you're probably looking from the compressor in the first place. Here are four basic steps to set up a compressor by ear:

1. Select The Correct Compressor For The Track

Different compressors have naturally slow or fast response times and that should be taken into account when selecting a compressor. For instance, an optical-style compressor has a slow attack so it wouldn't work well on a kick or snare drum, yet it will do a bang-up job on mix elements that have naturally slow attacks like vocals or strings. Likewise, VCA-style compressors have very fast attack times and a wide variety of parameters available so they work well on drums and percussion as well as the mix buss.

2. Select The Correct Compression Ratio For The Track

Of all the compression parameters, this one gets overlooked the most, which is odd because it's actually pretty easy to set up. If you want more punch from the track, set the Ratio control low. If you want more control, set it higher. In other words, if you want to maintain the snap of a kick or snare yet add some punch, set it at 2:1 (or even 1.5:1) so some of the transients get through.

If you want to make sure that the signal more or less stays at one level, set the ratio higher, like 4:1. Some engineers love a solid bass that doesn't

dynamically move, so they'll set the ratio at 8:1 or more (10:1 and higher is considered limiting).

TIP: Once again: set the Ratio low for punch; higher for control.

3. Select The Correct Compressor Dynamics For The Track

Again, this seems to be a mystery for some but if you set this up correctly in the beginning of your mix, your entire track, not just the channel you're working on, will feel a lot better. The idea is to get the compressor to breathe with the tempo of the song and the way you do it is to start by setting the Attack as slow as it will go and the Release as fast as it will go. Gradually decrease the Attack time until the high frequencies of the track begin to dull, then back it off a little.

Then increase the Release time until the sound just begins to die at the end of the beat before the next downbeat (easiest to use the snare to set this up). In other words, you're trying to set the compressor up so it breathes with the track.

4. Make Sure The Compressor Is Actually Helping The Track

It's really easy to think that the adjustments you're making with the compressor are helping the track when it's louder than the original signal, but our ears are easily deceived by anything that appears louder to us. We automatically think it sounds better. This is why it is critical to perform an

A/B comparison between the original signal and the compressed signal at exactly the same level.

Use the Gain control to adjust the compressed signal so it's identical in level to the bypassed signal, then listen to it in the mix to see if it's actually giving you the sound you want. After you're happy with the sound, you can adjust the Gain control all you want.

These four compressor tweaks can make a big difference in how the processor reacts with the track. Not only will it sound better, but the entire mix will feel better with all the compressors breathing at the same tempo.

TIP: The idea of setting the compressor's attack and release is to make it breathe with the pulse of the song.

"I get the bass and drums so they just start to pump to where you can actually hear them breathing in and out with the tempo of the song. What I'll do is put the drums and bass in a limiter and just crush the hell out of it. Then I'll play with the release and the attack times until I can actually make that limiter pump in time with the music, so when the drummer hits the snare, it sucks down and you get a good crest on it. When he lets go of the snare, the ambience of the bass and the drums suck and shoot back up again. You can actually hear a [breathing sound] going on that was never there before. It really was there; it's just that you're augmenting it by using that limiter."

—Lee DeCarlo

Setting The Compressor By The Numbers

There's another way to set up a compressor in time with the tempo of the track and that's by using the numbers that correspond to the note denominations of its tempo. In other words, setting the Attack and Release to different combinations of ¼, ⅛, 1/16th notes and so on.

To do this, first determine the tempo of the track. If it's not known you can use any of the smartphone BPM calculator apps available, or open up a delay plugin and tap in the tempo (more on how to do this in Chapter 10).

Once you've determined the tempo, you can either look at the delay times when set to a note denomination (¼, ⅛, etc.) or look it up on the BPM to Delay chart in the addendum of this book.

Here's a chart that will give you the starting times to try for a variety of mix elements. Remember, these are just starting points. Try the next higher or lower note denomination to see if it works better.

Mix Element	Ratio	Attack	Release
Kick	2:1	1/64th note	1/16th note
Snare	2:1	1/64th note	1/16th note
Drum Subgroup	2:1	1/64th note	1/16th note
Bass	12:1	1/32nd note	1/16th note
Mix Buss	2:1	1/16th note	1/16th note
Vocal	4:1	1/6th note triplet	¼ note
NOTE	Use higher ratios for more aggression	Times are starting places. Try other note denominations	Shorter attack = more aggression Longer releases = more aggression

You’ll notice that the number of mix elements on the chart is limited. That’s because there’s a wide variety of keyboard and guitar sounds possible and they might require different timings depending on the arrangement. Also, many times compressors with no Attack and Release controls are used like the LA3-A or dbx 160-style of compressors. The same can be said for vocals as well, where the LA-2A style compressors are a popular choice, as is the 1176-style, which controls that work backwards from all other compressors on the market (setting the Attack to 3 and the Release to 5 are popular settings).

What’s The Right Amount Of Compression?

The amount of compression added is usually to taste but generally speaking, the more compression, the greater the effect will be and the more likely it will change the sound of the mix element. Compression of 6dB or more is meant for controlling the dynamics of a track rather than for changing its sonic quality, as often just a dB or two is all that’s needed to change the track’s color or bring it forward in the mix.

That said, it’s not uncommon for radical amounts of compression to be used sometimes. Ten or even 20dB is routinely used for electric guitars, room mics, drums and even vocals. As with most everything else, it depends on the song, the arrangement, the player, the room, and the gear.

“There are times when there’s singing when they’re not in compression at all, but if my limiter hits 15 or 20dB of compression and I don’t hear it, I don’t think about it for an instant more.”

—Nathaniel Kunkel

TIP: The higher the compressor ratio control is set, the more likely you'll hear the compressor or limiter work. The more compression that's added, the more likely that you'll hear it.

Parallel Compression

Parallel compression is a trick that mixers have been using ever since the '70s to make a track sound punchy in a very natural way. The trick involves sending the signal to another channel via a buss or aux on a DAW mixer, or through a mult (a patchpoint on the patchbay that splits the signal) on a hardware console, and by then adding a compressor to the second channel only (see Figure 8.15).

The compressed channel is then brought up in level so it's just under the original non-compressed channel. This gives the track a feeling of control with added punch without sounding too squashed. Keep in mind that parallel compression works well with just about any individual instrument or vocal, and if you're working in a DAW, you're limited only by the number of tracks that you want to deal with.

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Figure 8.15: Parallel compression

"On this mix right now there's a parallel compressor on the kick and snare, then there's another just on the snare. There's a stereo one on the toms and overheads, a mono one on just the dirty bass (this song has three basses), a stereo one on the guitar and vocals, and then a couple of different ones just for the lead vocal, one that's sort of spitty and grainy and one that's sort of fat. That changes from mix to mix."

—Andrew Scheps

The New York City Compression Trick

When I wrote the first edition of this book in 1998, I noticed that a surprising number of mixers based in New York used a technique that I call the "New York City Compression Trick." Since then, mixers no longer limit their work to one territory and the information on the technique is everywhere, but it was a trick that made mixes done in NYC back then uniquely more punchy and aggressive sounding The technique came about because in the early days of recording, back when there were only a few compressors in each studio, this was a way to compress the entire rhythm section with just a single unit. Even if you don't mix in NYC, once you try the New York City Compression Trick you just might find yourself using this technique all the time because it's indeed a useful method to make a rhythm section rock.

Here's the trick:

Route the drums, and even the bass, via a send or buss to a stereo DAW channel with the compressor inserted.

Adjust the threshold of the compressor so there's at least 10dB of compression with a 4:1 ratio and fast attack and medium release. (Use even more if it sounds good.)

If you're on a real console, return the output of the compressor to a new pair of input channels. (No need to do this if you're using a DAW since you already have a stereo aux channel with the compressor inserted on it set up.)

Add some high frequencies (6 to 10dB at 10kHz) and low frequencies (6 to 10dB at 100Hz) to the compressed signal. (This is optional, as you'll still get the sound without the EQ.)

Bring the fader levels of the compressor channel up until it's tucked just under the present rhythm section mix to where you can just hear it.

The rhythm section will now sound bigger and more controlled without sounding overly compressed.

One of the cooler things about many of the latest generation of compressor plugins is that they have Blend or Mix controls which lets you set up parallel compression without having to go through some of the gyrations noted above. Although you still have to use some of the same routing for the New York City Compression trick, simple parallel track compression is now a snap.

“What I will do a lot is buss my drums to another stereo compressor and blend that in just under the uncompressed signal. Sometimes if everything sounds good but the bass and kick drum aren’t locked together or big enough to glue the record together, I’ll take the kick and bass and buss them together to a separate compressor, squish that a fair amount, and blend it back in. I’ll add a little bottom end to that if the record still isn’t big enough on the bottom. This helps fit the bass and kick lower on the record and gets it out of the way of the vocal.”

—Joe Chiccarelli

TIP: Remember that the real New York City Compression Trick is not just parallel compression on the drums. There’s an EQ in there, and sometimes even the bass is added.

Compression On Individual Instruments

Back in the early days of recording, a studio was lucky to have more than a couple of compressors available which meant that you had to be judicious in how they were used. This all changed as recording became more and more sophisticated, to the point where eventually large-format consoles came with dynamics built into every channel.

Today, when it comes to adding dynamics in a DAW, the limiting factor is the processing power available in the computer. While it may not be necessary or even desirable to add compression on every track in a session, sometimes tracks can benefit from just a slight touch. Let’s look at the effect compression can have on individual music tracks.

"To me, the key to compression is that it makes the instrument sound like it's being turned up, not being turned down. If you choose the wrong compressor or you use it the wrong way, then your stuff can sound like it's always going away from you. If you use the correct compressor for the job, you can make it sound like, 'Man, these guys are coming at you.' It's very active and aggressive."

—Ed Seay

A Drum Compression Primer

One thing that most modern mixes call for is a punchy drum sound and this can be achieved with compression, although a great acoustic drum sound, a great drummer, and a great recording are certainly all major components to the sound. It would be wonderful if every drummer hit every beat on the kick and snare with somewhat the same intensity, but unfortunately that doesn't even happen with the best session drummers you can find.

Some intensity changes are both proper and natural in music but when the intensity noticeably changes from beat to beat, the pulse of the song can feel erratic and sometimes even a slight change in level can make the drums feel a lot less solid than they should be.

Compression works wonders to even out those erratic hits and helps to push the kick and snare forward in the track to make them feel punchier. Let's take a look at how to do that with the drums.

Compressing The Kick And Snare

The most common question that arises when compressing either the kick or the snare is, “How much is enough?” This depends first and foremost on the sound of the drum itself and the skill of the drummer. A well-tuned drum kit with new heads that sounds great in the room should record well, and a reasonably good drummer with some studio experience usually means that less compression is needed because the hits are fairly even to begin with. A heavy-hitter drummer might mean that the drums are already punchy without any processing added.

Even a great drummer with a perfectly tuned kit can sometimes benefit from a bit of compression though, and as little as a dB or two can work wonders for the sound in many situations. With only that amount, the setup of the compressor, especially the attack and release. is a lot less demanding. .

TIP: A ratio setting of 2:1 sometimes provides the most punch as it lets more of the transients through.

But sometimes you need the kick or snare to cut through the mix and seem as if it’s in your face, and that’s when 3- to 6-dB or so does the job. It’s here that the setup of the compressor becomes more critical because you’re imparting its sound on the drum. Make sure to tweak the Attack and Release controls, as outlined earlier, as well as the Ratio control, and even try different compressors. You’ll find they all react differently, even with the same settings, so it’s worth taking the time to experiment.

TIP: If the attack is set too fast, the drum will sound less punchy, regardless of how much or how little compression you use.

Compressing The Room Mics

Room ambient mics are meant to add the “glue” to the sound of a kit and can generally benefit from a fair amount of compression, which means anywhere from 6 to 10dB. In fact, many mixers prefer the room sound to be extremely compressed, with way more than 10dB being the norm.

The problem is that the more compression you use, the more the ambience of the room is emphasized. That’s okay if the room tracks were recorded in a great sounding space, but if the recording environment has a lot of acoustic reflections and the ceiling is low, you may be emphasizing something that just doesn’t add much to the track.

TIP: If the ambience on the room mic tracks sounds bad, set the attack time so it’s much shorter than usual to cut off the sound of the initial drum transient, and keep the release time short so the ambience isn’t emphasized, then tuck the room tracks in just under the other drum tracks.

Note that regardless of how good the room mics sound, the more of them you use, the less space there will be for the other instruments in the track. The more mix elements there are, the more you’ll have to back them off. Sad but true, but there’s only so much sonic space to any mix.

The 1176 Nuke Setting

One technique for getting outrageous sounding room mic sounds is to use an 1176-type compressor on what's known as the "Nuke" or "British" setting. On the hardware version, if you push all four Ratio buttons in at the same time, the compressor goes into a wild new mode with some over-the-top compression (see Figure 8.16). The meter pins, the Attack and Release control no longer work, and you'll hear a sound that you'll either love or hate. Needless to say, the designers of the unit never thought that users would be pushing all the buttons in at the same time on the original unit, so the effects aren't documented.

It turns out that the result is perfect for room mic tracks as it emphasizes the sound of the room and makes it feel a lot larger than it might actually be. This setting can be used for parallel compression as well.

1176-type plugins now add this setting either as a preset, or make it accessible by holding shift and selecting all the Ratio buttons. The 1176 is a wonderful compressor but this effect really sets it apart from the rest of the field (although the Empirical Labs Distressor also has a Nuke mode).

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Figure 8.16: 1176 in Nuke mode

The Phil Collins Drum Sound

Another example of extreme compression that was discovered accidentally is the tom fills at the beginning of Phil Collins' "In The Air Tonight". This instantly recognizable percussion (phrase) has become an iconic musical statement, but its unique sound was due to a quirk of the SSL console.

SSL's have a feature known as the "Listen Mic" which is a mic connected to the studio that allows the engineer to hear what the players are saying when not recording. The mic is connected to a separate channel with a very fast 100:1 limiter that makes sure that everyone on the studio floor can be heard.

When the Listen Mic was accidentally engaged during a run-through of "In The Air Tonight", a new sound was born.

SSL has now incorporated this feature on many of its plugins to make it easy to get that same effect, and other software developers like Korneff Audio also make a plugin version (see Figure 8.17)

Graphical user interface, application Description automatically generated

Figure 8.17: Korneff Audio Talkback Limiter

Courtesy Korneff Audio

Compressing Vocals

Most singers aren’t able to sing every word or line at the same level, and some words get buried in the mix as a result. Compression helps to even out the level differences so it’s easier to hear every word.

Depending on the vocal style, the song and arrangement, the mic technique and any number of other factors, the amount of compression required on a vocal can vary a great deal. Some vocalists are so consistent that only a dB or two is needed, but it’s not uncommon to use as much as 10dB or more on some vocal tracks. Here’s how to set up the compressor on a vocal:

Solo the vocal and insert a compressor on its channel. Begin with the Ratio set to 4:1, but experiment with higher ratios if you’re not getting the sound or control that you’re looking for. You can also use a 2:1 ratio with more compression to make it sound smoother with less of the sound of the compressor.

Try setting the Threshold to where there’s about 2dB of compression and notice if there are any words that are less understandable than others. If so, increase to about 6dB and listen again. Don’t be afraid to increase the compression, but understand that the more compression you add, the more you’ll hear it work and the more color it will add.

TIP: Sometimes when a lot of compression is being added, inserting an additional compressor in the signal path and splitting the amount of work between the two sounds better.

Set the Attack and Release controls as described previously to breathe with the track, but also experiment with extreme settings. If there is a Knee

control, set it to a soft knee so that the compressor gradually begins to compress when it hits threshold.

Bypass the compressor and listen to the level, then add Makeup Gain to make the before-and-after compression levels the same.

With the Attack and Release parameters set, unsolo the lead vocal and listen to what it sounds like in the track, then tweak as necessary.

Alternative Setup

Another way to set up the compressor on a vocal is to go to the place in the song where the vocal is at its quietest and set the compressor so there’s zero compression at that point. As the vocal gets louder, it should stay the same level since more compression will be applied.

Because you likely will be adding much more compression than with the previous method, properly setting the Attack and Release controls is critical for a smooth sound without introducing unwanted audio artifacts.

TIP: Compressing the vocal more may not alleviate the need for automating the track, because even though the vocal may stay at a steady level, the density of the arrangement may change around it.

Compressing Loops

Much of modern music is derived from samples, beats and loops, but a danger lurks when compressing anything from a loop library, because most of the components may be heavily compressed already. A potential byproduct of using additional compression on a loop is that it could change its groove, which won't be desirable in many cases. That said, sometimes just a few dB of compression can handle any peak that might exist and allow it to sit better in the mix.

Compression On The Mix Buss

Along with compressing individual tracks, many engineers place a stereo compressor across the mix buss to affect the entire mix as well. This practice started during the late '70s, when artists began asking why their mixes sounded different in the studio from what they heard on the radio or when their record came from the pressing plant. (This was back in the vinyl days.) Indeed, both the radio and record did sound different because an additional (layer?) round or two of compression was added in both mastering and broadcast.

To simulate what this would sound like in a listener's ear — via radio or LP — mixing engineers began to add a touch of compression across the mix buss. The problem was, everybody loved the sound, so now it's common for mixes to have at least a few dB of compression added to the stereo mix, even though it will probably be re-compressed again at mastering and yet again if played on the radio or television.

“Compression is like this drug that you can’t get enough of. You squish things and it feels great and it sounds exciting, but the next day you come back and you’re saying, ‘Oh God, it’s too much,’ so I’ve been trying to really back it off, especially with stereo buss compression.”

—Joe Chiccarelli

When it comes to compressing the mix buss, not all compressors are up to the task. Because only a few dB of compression may be all that’s added (although it can often be a lot more), the compressor itself actually adds an intangible sonic quality. Among the many favorites are the Fairchild 670 (Figure 8.18), the Neve 33609 (Figure 8.19) and the most famous of them all, the SSL Buss Compress (Figure 8.20).

“Generally, the stereo buss itself will go through a Fairchild 670 (serial #7). Sometimes I’ll use a Neve 33609 depending on the song. I don’t use much, only a dB or 2. There’s no rule about it. I’ll start with it just on with no threshold, just to hear it.”

—Don Smith

Graphical user interface Description automatically generated

Figure 8.18: Universal AudioFairchild 670 plugin

Courtesy Universal Audio

Graphical user interface Description automatically generated

Figure 8.19: Neve 33609

Courtesy Neve

The SSL Mix Buss Compressor

The sound of many great records from the ’80s and ’90s owes thanks to the built-in mix buss compressor on an SSL console (Figure 8.20). This is an aggressive compressor with a very distinct sonic signature. Some have even gone so far as to call the SSL compressor’s “In” button (meaning it’s present in the signal path) the “Good” button, because it makes everything sound better.


SSL G-COMP
0 4 8 12 16 20
dB
COMPRESSION
IN
THRESHOLD
0
-20 +20
dB
MAKE UP
0
– +
dB
ATTACK
.1 .3 1 3 10 30
ms
RELEASE
.1 .2 .4 .8 1.6 AUTO
s
RATIO
1.5 2 3 4 5 10
HPF
OFF 30 60 105 125 185
Hz
STEREO BUS COMPRESSOR

Figure 8.20: SSL buss compressor

Courtesy Solid State Logic

If you happen to get an opportunity to work on an SSL console (of any vintage—they all have a mix buss compressor), the outboard rack-mount version, the version by Alan Smart, or any of the various plugin emulators, here's the time-honored setting to use as a starting point.

TIP: The typical SSL buss compressor settings are:

* Attack: .1 or .3

* Release: Auto

* Ratio: 4:1

* Threshold: To taste

Generally, you'll find that well-seasoned mixers use the buss compressor to add a sort of "glue" to the mix so the instruments fit together better, but this doesn't necessarily mean it requires a great deal of compression. In fact, sometimes only a dB or two of gain reduction is added for the final mix. That being said, many mixers will also offer their clients (artists, band members, producers, and label execs) a more compressed version to simulate what it will sound like after it's mastered. This "client mix" is achieved by using a signal path across the mix buss that's similar to what a mastering engineer would use—that is, a compressor fed into a limiter at the end of the chain — to raise the level to more closely resemble that of a mastered release (see Figure 8.21).

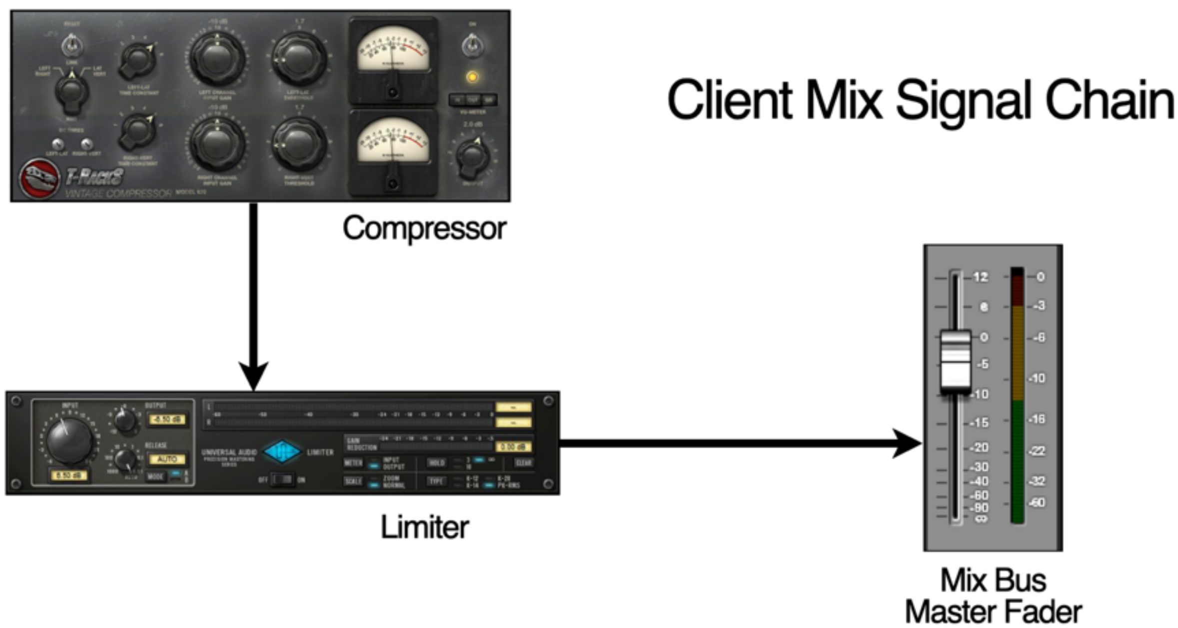

Figure 8.21: The master buss signal path for a client mix

Because clients can get used to hearing the “client mix,” it’s easy for the heavy buss compression to get out of hand. One of the problems with compressing too much is that it leaves the mastering engineer with a lot less room to work, and in the case of a track that’s “hyper-compressed,” it virtually eliminates the ability for the mastering engineer to be of much help at all. That’s why care should be exercised anytime compression is added to the mix buss. (See Chapter 13, “The Master Mix,” for more on delivery to mastering.)

That said, there are two times during the mix that you can insert your buss compression: when you first start the mix or as you near completion. While the two choices might not seem all that different, you will get a slightly different result from each.

At The End Of The Mix

If you wait to insert the buss compressor until later in the mix, the compressor settings can be less aggressive, because a fair amount of compression may have been already inserted on the individual tracks. If you choose to wait until later in the mix, usually the best time to insert it is at the point where most of your elements and effects have already been added and it's now time to concentrate on final balances.

One advantage of inserting buss compression toward the end is that if you don't like the sound, you can easily substitute a different compressor or even eliminate it all together.

TIP: If you're using the right compressor, it may take only a dB or two of buss compression to make a sonic difference to where the mix is bigger sounding.

At The Beginning Of The Mix

The other approach is to insert the buss compressor right at the start of the mix and build your mix into it. Because this affects the dynamics of the mix

right from the beginning, mixing this way might take a little getting used to but it does have its advantages.

First, the mix comes together a little quicker since it has that “glue” almost right from the start. Second, you may find yourself using a little less compression on individual tracks. This has the secondary benefit of giving you greater control of the overall compression of the mix. If you feel like there’s too much, it’s easy to back off on the buss compressor to the point where you or your client feel better about the results, whereas if you added it toward the end, sometimes the only way to dial back the total compression of the mix is to tweak the individual instrument compression, which can take quite a bit of time and rebalancing.

The third advantage is that the buss compressor tends to even out the levels of the individual instruments a lot, so you might not need to automate the fader levels as much. The downside of doing it this way is that if you decide you don’t like the sound of the compressor, the overall sound and balance of the mix can change significantly when you insert a different one.

Finally, when starting with the buss compressor in the signal path from the beginning of the mix, you may find that you’re using somewhat more buss compression than if it was introduced toward the end of the mix.

Compression Techniques

Here are a number of techniques often used when adding compression to particular mix elements. Don’t limit yourself to just these examples, as the

settings can just as easily work for other instruments in other situations as well.

For Snare

To compress the snare drum to get more sustain so it sounds fatter, go to a part of the song where the drums are playing straight time and adjust the release time to elongate the sound. Set it so it doesn't begin to attenuate until the next snare hit.

To remove snare leakage from the overheads without making the hi-hat sound too ambient, compress the overhead mics and key them from the snare by sending the snare signal into the sidechain input of overheads' compressor.

Instead of adding more high-end EQ to the snare, try compressing it instead, but be sure that the attack is set fairly long so that the initial transient gets through and is not compressed. This will allow you to elongate the snare drum's duration and create the illusion that it is brighter.

"What I do a lot is take a snare drum and go through an LA-2, just totally compress it, and then crank up the output so it's totally distorted and edge it in a little bit behind the actual drum. You don't notice the distortion on the track, but it adds a lot of tone in the snare."

—Ed Stasium

If a sample is being added to the snare (see Chapter 12 for more on how that's done), compress the original snare and not the sample, because the sample has probably been processed already. This works when replacing other drums as well.

For Kick

For a punchy kick, set the attack time slow enough to let the initial attack through, and set the release time so that it just begins to die at the next kick beat. Increase the ratio to add more punch.

For a floppy sounding kick (either with or without a front head), set the release time shorter than normal to deemphasize the end portion of the kick signal.

Synthesized kicks, like from the famous 808, typically have a lot of low end on them that sounds better when it's controlled. Try some parallel compression by sending it to another channel that has a compressor inserted, compressing it with a very fast attack and release, and mixing the compressed signal back in with the original to help beef it up.

For Room Mics

Compress the room mics by 10 to 20dB to increase the room sound with a slow attack time and the release time fairly long or timed to the track. If the acoustics of the room sound good to begin with, it will sound tighter and cleaner than an outboard reverb.

For Bass

Set the threshold of a compressor to its highest ratio (even if it's infinity:1) with at least 3 or 4dB of gain reduction. This will keep the bass solid with the level constant in the mix.

With a bass track that has no definition to the notes, try an 1176 with the attack set to around the middle and the release set to around 3 or 4 o'clock with an 8:1 ratio and 4 to 6dB of gain reduction. This helps put a front end on the notes that may not have been articulated properly when the part was recorded.

With a bass track where the notes don't sustain long enough, increase the release to where the longest notes don't die until the next one sounds.

If a bass player is using a pick, it may create high midrange transients that might need to be limited even before compression is added. Insert a limiter first and set the attack and release times fairly fast; then insert a compressor set from medium to slow on both attack and release and a ratio of 4:1.

On some rock bass parts played with a pick, a multiband compressor across the bass can achieve a more even level without sounding compressed.

For Vocal

A good starting point for a lead vocal is a 4:1 ratio, medium to fast attack and medium release, and the threshold set for about 4 to 6dB of gain reduction.

When a single compressor or limiter just isn’t enough for a troublesome vocal, try an 1176-style plugin set to fast attack (all the way clockwise, which is backward from other compressors), the release set to medium (5), and the ratio set to either 8 or 12:1 to clip the peaks by 4 to 5dB. Feed the output of the 1176 into an LA-2-style plugin set for gentle gain riding of 2 to 3dB.

As an option to above, if you insert a second 1176, then set the first one on fastest attack and an 8:1 ratio, and the second on a slower attack and a 4:1 ratio. With both set for about 4 to 6dB of compression, the vocal will be silky and smooth, yet still in your face.

For Piano

If you liked the early Elton John piano sound, insert an LA-2A or similar compressor plugin (use a stereo plugin if the track is stereo) and compress the signal at least 10dB. Next, add a Pultec or similar style equalizer. Boost at 14kHz and 100Hz to taste. The effect should be a shimmering sound where the chords hold and seem to chorus.

For Guitar

Higher ratios of compression around 8:1 or 10:1 sometimes work well, with the threshold set so that the guitar cuts through the track. Attack and Release should be timed to the pulse of the song.

“I may go 20:1 on a [UREI] 1176 with 20dB of compression on a guitar part as an effect. In general, if it’s well recorded, I’ll do it just lightly for peaks here and there.”

—Don Smith

When a guitar is recorded with both close and distant mic tracks, you can use this trick to get guitars to sound big, yet stay out of the way of the vocal. Pan the distance mic in the same direction of the close mic and then attach a compressor across the distance track keyed off the lead vocal. When the lead vocal is present, the ambience is decreased. When the lead vocal stops, the ambience returns to full level. As a variation, try panning the distance mic track to the opposite side of the close mic track.

With doubled guitars, pan them medium left and right, then send the same amount of both to a compressor on a separate track, either via an aux send or

a buss. Pan the output of the compressor track to the center and bring it up just underneath the original tracks. Now the guitars don't have to be as loud as before to still have presence.

If you notice a guitar getting lost in places, try compressing with a ratio of 2:1 to 4:1 with medium attack and release times. This will allow the rhythmic transients to get through somewhat uncompressed while boosting the sustain of the guitar sound.

With rock guitars, the idea is to have them big and in-your-face. This is accomplished by first limiting the transients from the signal with a ratio of 10:1 or higher. Be careful to set the release so that any sustaining parts of the signal return to unity gain before the next transient.

Try an LA-3A-style plugin on an insert of the guitar track. This compressor has no Attack or Release controls but automatically works well on the dynamics of a guitar. It can be used on clean or distorted guitar parts. Just a dB or two of compression will bring the guitar forward in the mix.

Using A De-Esser

Sibilance is a short burst of high-frequency energy where the S's are overemphasized. This comes from a combination of mike technique by the vocalist, the type of mic used, and heavy compression on the vocal track. Sibilance is highly undesirable, so a special type of compressor called a de-esser is used to suppress it (see Figure 8.22).

Figure 8.22: Logic Pro X DeEsser 2

Courtesy Apple, Inc

Most de-essers have two main controls, Threshold and Frequency, which are used to compress only a very narrow band of frequencies, anywhere between 2kHz and 10kHz, to eliminate sibilance. Modern software de-essers are much more sophisticated, but the bulk of the setup still revolves around fine-tuning those two parameters. One frequently used additional feature is a Listen button that allows you to solo only the frequencies that are being compressed, which can be helpful in finding the exact band of offending frequencies.

To use a de-esser, do the following:

Insert the de-esser on the vocal channel and solo it.

Lower the Threshold control until the sibilance is decreased, but you can still hear the S’s. If you can’t hear any at all, then you’ve lowered the Threshold too far.

Scan the available frequencies with the Frequency control until you find the exact spot where the S’s are most offensive, then adjust the Threshold control until the they sound more natural.

Un-solo the vocal and listen in context with the track. Be careful that you don’t completely eliminate all the S’s, because that makes it sound unnatural.

Sometimes the sibilance occurs in two or more places of the frequency spectrum, so you may have to use a multiband de-esser or multiple plugins to be effective.

TIP: When using the Listen feature, remember that the audio you’re hearing isn’t in the signal path, just the sidechain. Don’t forget to disengage Listen when you’ve found the correct frequencies.

Using A Gate

Like the de-esser, a Gate can sometimes consist of just a few controls, principally the Threshold, Range, and sometimes a Hold or Release control (see Figure 8.23). Range sets the amount of attenuation after the signal drops below the threshold when the gate turns off.

Sometimes when gating drums, the Range control is set to attenuate the signal only by about 10 or 20dB. This lets some of the natural ambience remain and prevents the drum from sounding choked. The Hold control keeps the gate open for a defined amount of time, and the Release control sets how quickly the gate closes again.

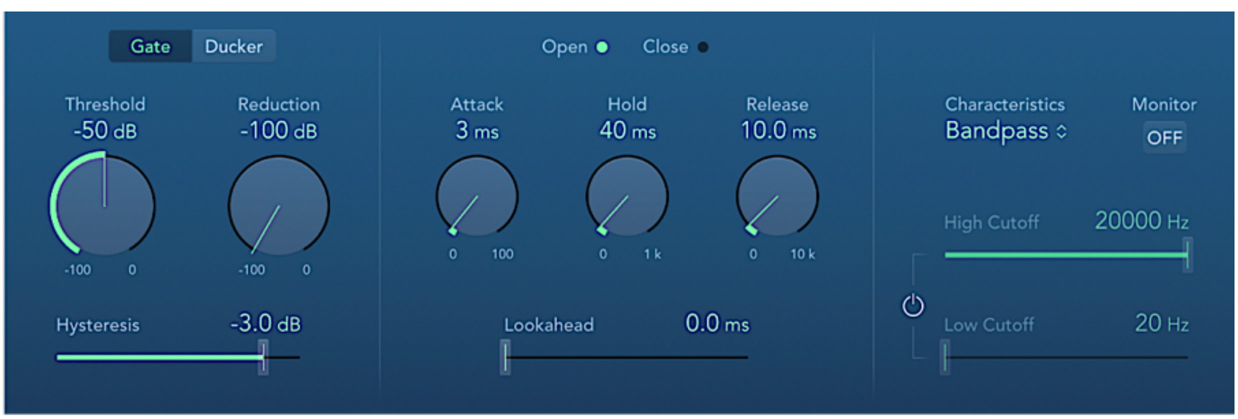

Figure 8.23: A typical noise gate

Courtesy Solid State Logic

We’ll use the snare drum as an example of how to set up a gate, but the same technique is used to set up a gate on any drum, instrument, or vocal.

Insert the gate on the snare channel and solo it.

Raise the Threshold control until you can hear the snare drum hit, but there’s no sound in between the hits.

If the snare sounds unnatural or cut off, raise the Threshold a bit to see whether that improves the sound.

If the snare still sounds unnatural or cut off, try increasing either the Hold or the Release control or both.

Try adjusting the Range control so the snare is attenuated by 10 dB between hits to hear whether it sounds more natural or improves the tone of the overall drum kit.

If the gate chatters (continually opens and closes quickly), try fine-tuning the settings of the Threshold and Release controls. This may require some experimenting, so be patient.

Try timing the Release control so that the gate breathes with the pulse of the song.

TIP: Gates are finicky because the dynamics of the signal are usually constantly changing. Inserting a compressor in the signal chain before the gate improves the performance greatly.

Gating Techniques

Here are some techniques often used when gating a mix element. Gates usually take time to set up properly, so take your time and don't be afraid to experiment.

For Snare:

This simple technique allows a different effect to be placed on the snare during harder hits and prevents leakage to the effect during tom hits, hi-hat, or stray kick-drum beats. Duplicate the snare to another channel and insert a gate on this new channel. This gated channel is generally not sent to the main mix (although it can be), but is used primarily as an effects send to a reverb. By adjusting the threshold, you can have more control over how the signal is sent to the effects unit.

Another way to make the snare feel as if it's breathing with the track is to copy the snare onto a second track and insert a gate on only that channel. Time the release so it cuts off right before the next snare hit during the main part of the song when the snare is just playing time. Bring the gated snare channel up just underneath the main snare track.

To make the snare drum sound bigger, gate either room ambience or reverb and trigger it from the snare by sending the snare signal to the trigger/key input of the gate.

For Drums:

When gating toms, set the Range control so it attenuates the signal only about 10 or 20dB. This lets some of the natural drum ambience remain and prevents the drums from sounding choked or unnatural.

To make the rhythm section feel tighter, feed the bass through a gate with only 2 or 3dB of attenuation when the gate is closed. Trigger the gate from the kick drum so that the bass is slightly louder on each kick drum beat.

For times when the groove doesn’t quite lock or the bass player is playing on top of the beat and the drummer is laying back, insert a gate on the bass channel and key it from the kick drum. The bass will only play when the kick drum is played, so the rhythm section will now sound tight. Set the sustain and release so that it sounds more natural and less staccato, since the kick is a transient instrument and the bass is not, or leave it staccato if that works well.

The above technique can also be used to get more space and rhythm in a big chorus by gating the rhythm guitar track keyed from the hi-hat.

Key a gate placed across a synth or keyboard pad from the hi-hat to make the pad more rhythmic.

For tighter background vocals, patch the best phrased harmony line into the key insert of a gate across a stereo submix of the harmonies. This will ensure that all of the vocal entrances and exits will remain tight.

9

The Frequency Element: Using The Equalizer

Even though an engineer has every intention of making his or her recording sound as big and as clear as possible during tracking, production, and overdubs, it often happens that the frequency range of some (or even all) of the tracks is somewhat limited when it comes time to mix. This can be due to the tracks or loops being recorded in a different studio where different monitors or signal path are used, the sound of the instruments themselves, the processing used on a loop or sample, or the individual taste of the artist or producer. When it comes to the mix, it's up to the mixing engineer to extend the frequency range of those tracks if it's appropriate.

In the quest to make things sound bigger, fatter, brighter, and clearer, the Equalizer is the first tool of choice for by most mixers, but perhaps more than any other audio device, it's how it is used that separates the average engineer from the master.

"I tend to like things to sound sort of natural, but I don't care what it takes to make it sound like that. Some people get a very preconceived set of notions that you can't do this or you can't do that, but as Bruce Swedien said to me, he doesn't care if you have to turn the knob around backwards; if it sounds good, it is good. Assuming that you have a reference point that you can trust, of course."

—Allen Sides

"I find that the more that I mix, the less I actually EQ, but I'm not afraid to bring up a Pultec and whack it up to +10 if something needs it."

—Joe Chiccarelli

The Goals Of Equalization

While we may not think about it when we're doing it, there are four primary goals when equalizing:

To make a mix element sound clearer and more defined.

To create an aural depth of field by bringing mix elements in and out of focus.

To make the mix element or mix bigger and larger than life.

To make all the elements of a mix fit together better by putting each one in its own predominate frequency range.

Sometimes, just being aware of which outcome you're trying to accomplish at any given moment can help you get the results you're looking for quickly and easily, rather than just randomly adjusting parameters until you stumble upon something that might sound right.

The Frequency Bands And What They Do

Before we examine the various methods of equalization, it’s important to note specific areas of the audio frequency spectrum and how they affect what we hear. The audio spectrum can effectively be broken down into six distinct ranges, each one having an enormous impact on the total sound (see Table 9.1).

Table 9.1: The Audible Frequency Ranges

Range	Description	Effect
16Hz to 60Hz Sub-Bass	Encompasses sounds that are often felt more than heard and gives the music a sense of power.	Too much emphasis in this range makes the music sound muddy. Attenuating this range (especially below 40 Hz) can clean up a mix considerably.
60Hz to 250Hz Bass	Contains fundamental notes of the rhythm section.	EQing this range can change the musical balance, making it fat or thin. Too much boost in this range can make the music sound boomy.
250Hz to 2kHz Low Mids	Contains the low-order harmonics of most musical instruments.	Can introduce a telephone-like quality to the music if boosted too much. Boosting the 500Hz to 1000Hz

		octave makes the instruments sound horn-like. Boosting the 1kHz to 2kHz octave makes them sound tinny. Excess output in this range can cause listening fatigue.
2kHz to 4kHz High Mids	Contains speech recognition sounds such as “m,” “b,” and “v.”	Too much boost in this range, especially at 3kHz, can introduce a lisping quality to a voice. Too much boost in this range can cause listening fatigue. Dipping the 3kHz range on instrument backgrounds and slightly peaking 3 kHz on vocals can make the vocals audible without having to decrease the instrument level in mixes where the voice would otherwise seem buried.
4kHz to 6kHz Presence	Responsible for clarity and definition of voices and instruments.	Boosting this range can make the music seem closer to the listener. Reducing the 5kHz content of a mix makes

		the sound more distant and transparent.
6kHz to 16kHz Brilliance	Controls brilliance and clarity.	Too much emphasis in this range can produce sibilance on the vocals. 10kHz and above provides “air” and sense of realism.

di Gar Kulka, Leo. “Equalization - The Highest, Most Sustained Expression of the Recordist’s Heart.” Recording Engineer/Producer, Vol. 3, Number 6, November/December 1972

For those of us who have an easier time visualizing the audio spectrum in one-octave increments like those found on a graphic equalizer, here’s an octave look at the same chart (see Table 9.2):

Table 9.2: Graphic Equalizer Chart

Octave Band	Effect
31Hz	Rumble, “chest”
63Hz	Bottom
125Hz	Boom, thump, warmth
250Hz	Fullness or mud
500Hz	Honk
1kHz	Whack
2kHz	Crunch
4kHz	Edge
8kHz	Sibilance, definition, “ouch!”
16kHz	Air

Charts are fine, but here's another way to look at the various frequency bands more visually (see Figure 9.1):

Figure 9.1: The visual frequency spectrum

Courtesy of Captain Delanov.

Types Of Equalizers

It used to be there was only one type of equalizer with several variations on the basic theme. Thanks to some excellent software development and the emergence of artificial intelligence, EQ’s have progressed far beyond what an engineer could have even dreamt of a decade ago. Here are the equalization choices that every mixer can have at their fingertips today:

Static Equalizers

A Static Equalizer is a traditional EQ that adjusts a set of frequencies for the entire length of a song after it’s been mixed. This category of equalizer is broken up into several sub-categories.

Shelving EQ

In a Shelving Equalizer, all frequencies above or below a certain frequency are boosted or cut by the same amount (see Figure 9.2). This only applies to high and low frequencies and never to the midrange. Sometimes these frequencies are fixed (100Hz and 10kHz are popular frequency points), and sometimes these frequencies can be variable.

In guitar amps, hi-fi equipment and everyday audio playback gear, simple shelving EQ is commonly referred to as the tone controls or, more familiarly, the “bass” and “treble”adjustments. Typically, this type

of EQ has just a boost/cut control for the lows and highs, although a frequency selector or variable frequency control may also be included in more sophisticated models.

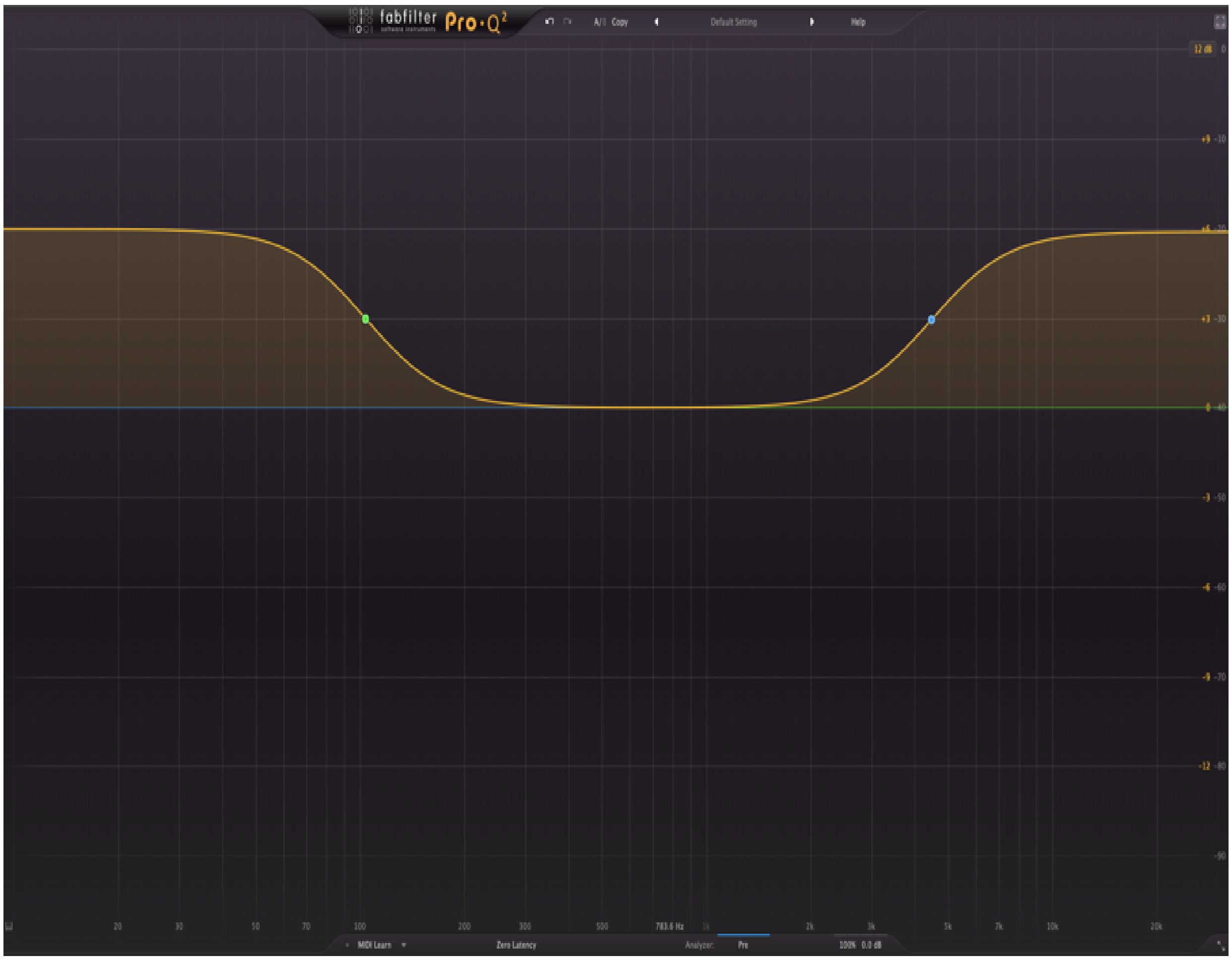

Figure 9.2: High and Low Shelving EQ Curve

Tilt EQ

A Tilt Equalizer is a variation of a simple Shelving EQ except that it tilts the frequency content of the audio signal by simultaneously boosting the treble and cutting the bass frequencies — or vice versa — to instantly make a track warmer or brighter (see Figure 9.3). Only one knob is needed to tilt the frequency response around a pivot frequency, This type of EQ is not seen very often, and in many cases (such as the Tonelux Tilt or Mixland TILT!) may be a separate, dedicated plugin.

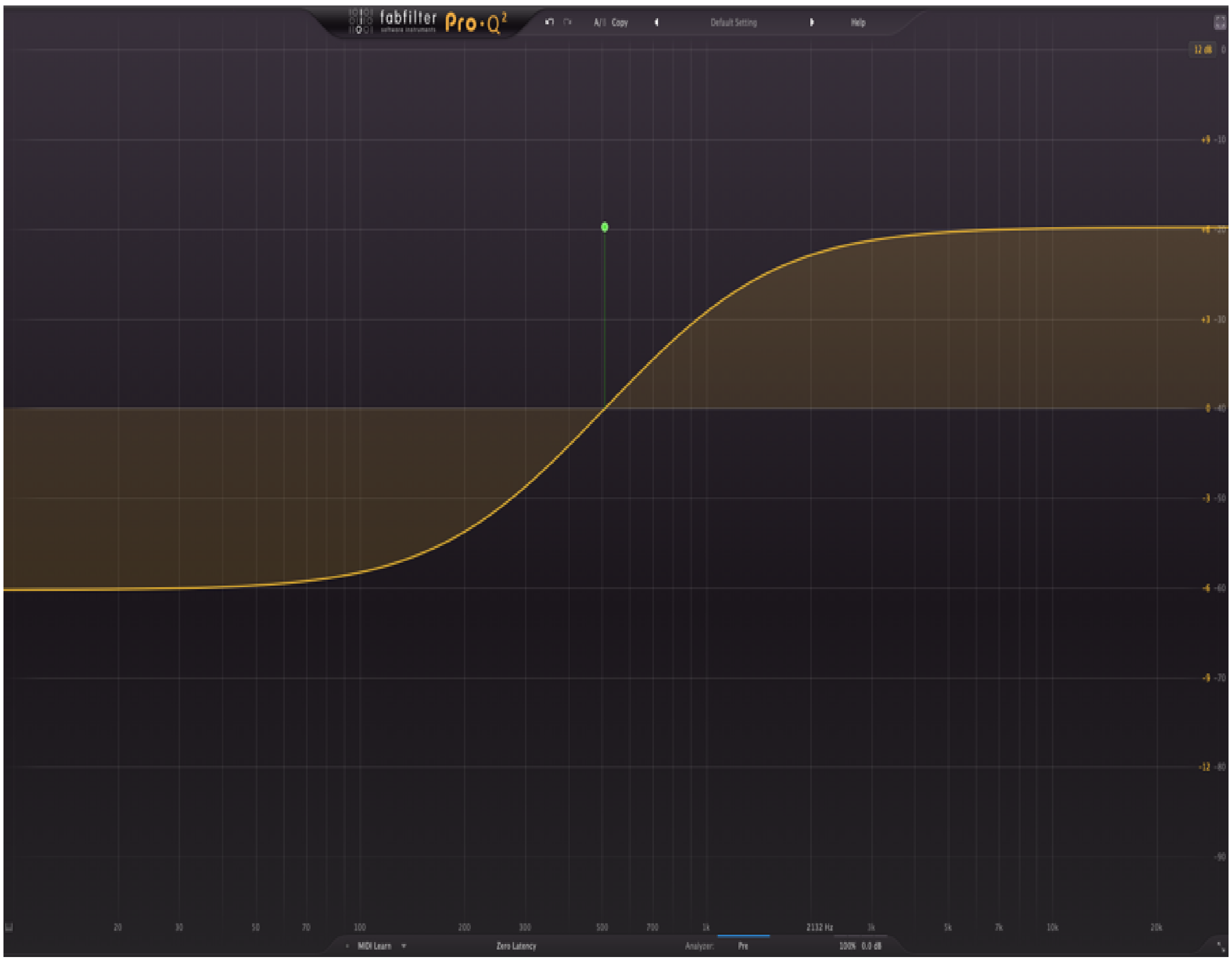

Figure 9.3: Tilt EQ at 500Hz

Peaking EQ

A Peaking EQ, sometimes called "bell EQ" because it's shaped like a bell (see Figure 9.4), extends the capabilities of a normal shelving-style EQ by adding a midrange control that boosts or cuts a band of frequencies around a center frequency. Sometimes the frequency is adjusted either by fixed increments or is continuously variable.

Usually, the Q or bandwidth of the Peaking EQ is fixed, but with some, like API's famous 550A and 550B or the Harrison 32C, have what's known as a proportional EQ. This means that the bandwidth narrows as more EQ is applied.

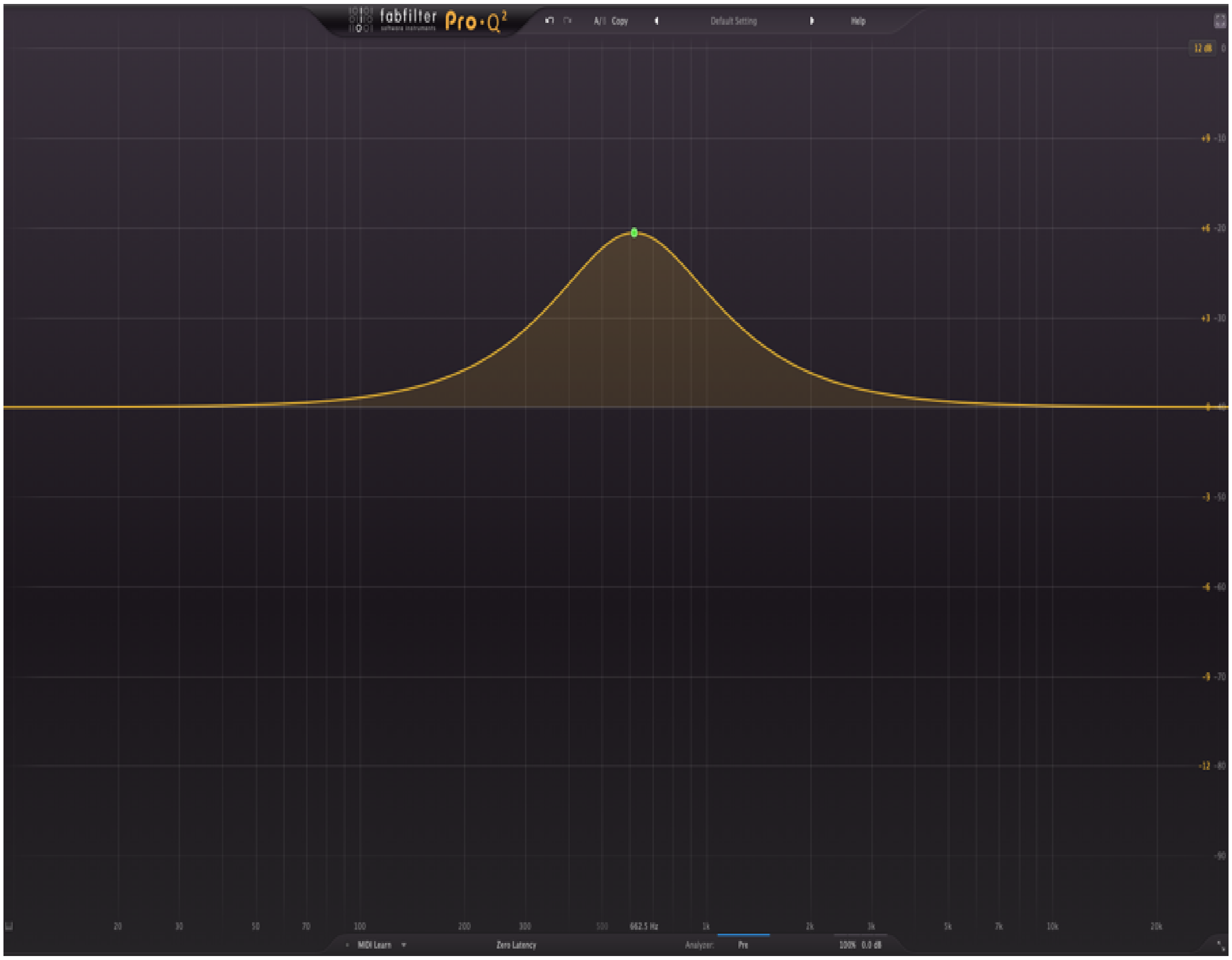

Figure 9.4: Peaking or Bell EQ Curve

Graphic EQ

A Graphic EQ is a connected series of peaking EQs that are centered around industry-standard frequencies. The simplest and probably most common is the octave graphic, which is a 5-band graphic representation with frequencies at 60Hz, 250Hz, 1kHz, 4kHz, and 16kHz, although those frequencies can change according to the application. A five-band

EQ typically covers: the sub-bass; bass; low-mid; high-mid; and brilliance frequencies; although not with great precision since the bands are so wide.

A ten-band graphic usually has its center frequencies at 31Hz, 63Hz, 125Hz, 250Hz, 500Hz, 1kHz, 2kHz, 4kHz, 8kHz and 10kHz.

One-third (1/3) octave graphic equalizers are used for room tuning and feedback elimination in live sound systems (see Figure 9.5). Their center frequencies are usually set to the ISO standard of 20Hz, 25Hz, 31.5Hz, 40Hz, 50Hz, 63Hz, 80Hz, 100Hz, 125Hz, 160Hz, 200Hz, 250Hz, 315Hz, 400Hz, 500Hz, 630Hz, 800Hz, 1kHz, 1.25kHz, 1.6kHz, 2kHz, 2.5kHz, 3.15kHz, 4kHz, 5kHz, 6.3kHz, 8kHz, 10kHz, 12.5kHz, 16kHz and 20kHz.

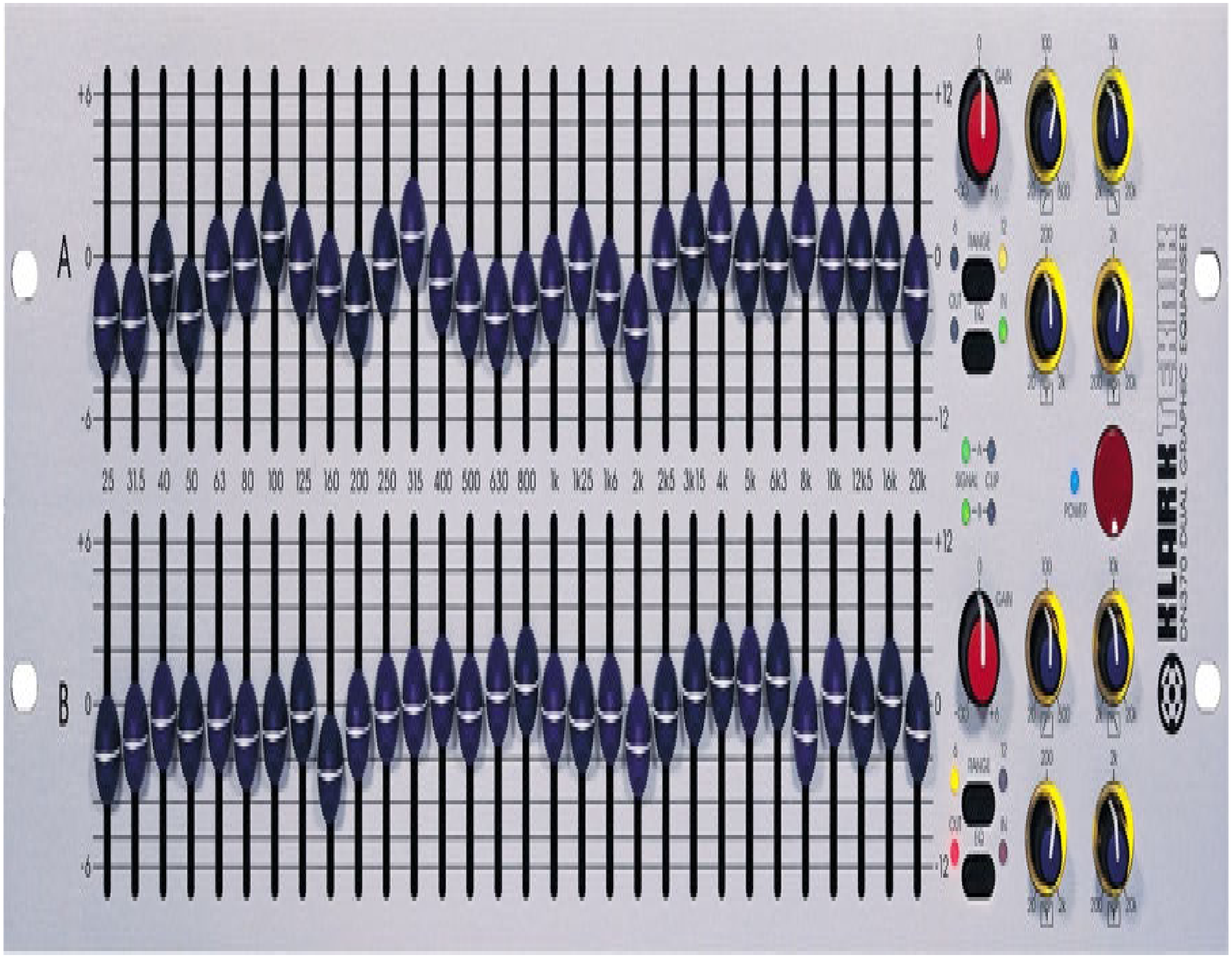

Figure 9.5: 31 Band Klark Teknik DN 370 Graphic EQ

Parametric EQ

A Parametric EQ takes a peaking equalizer one step further by adding a Q or bandwidth control to the boost/cut and frequency controls. The higher the Q (which stands for filter "quality") the more narrow the bandwidth. Low Q's of around 1 are better for boosting, while high Q's

around 10 are better for zeroing in on a narrow set of frequencies to manipulate (see Figure 9.6).

Some equalizers, like the well-known GML 8200 or Massenburg Design Works MDW EQ-5 (George Massenburg designed the first true parametric EQ in 1972), are fully parametric, meaning that all bands operate in this manner, while other models have the feature available for the midrange only.

Figure 9.6: Parametric EQ with wide Q boost and low Q cut

Dynamic Equalizers

A Dynamic EQ is a powerful tool that combines the capabilities of an equalizer with those of a multi-band compressor. The difference is that a dynamic EQ replaces the crossover filters of a standard multi-band compressor with the traditional EQ filter shapes that allow you to more precisely zero-in on a band of frequencies you want to process instead of the more general, less-specific frequency "areas". The result is that dynamic EQ allows you to adjust the frequency, gain and bandwidth of a band of frequencies, while adding additional compressor adjustments such as threshold, attack, and release. This greater control allows you to process audio in a way that many consider to be more musical than with static EQ.

Common Uses

A standard EQ is useful for when a track needs a constant boost or cut throughout the entire song, but sometimes an EQ problem only happens sporadically. That's when a dynamic EQ comes in handy. Here are a number of instances where dynamic EQ can be used to great effect:

Vocals

Sometimes a singer's voice will change according to how loud he or she is singing, for instance getting more shrill in a chorus as compared to a verse. Setting a dynamic EQ to dip a little 2k to 3kHz during those periods can mean the vocal will have the same tonal quality throughout the entire song.

Multi-Band De-Esser

Yet another use for a dynamic EQ is as a multi-band de-esser. Most de-essers are single or two-band at most, but sometimes there can be as many as four trouble areas causing sibilance throughout a song. Instead of using multiple de-essers, a dynamic EQ can be set so a band will compress at each sibilance frequency. Dynamic EQ's that work well in this application include the Waves F6-RTA and Fabfilter Pro-Q 3.

Toms

Many times, the tone of the snare drum and, for that matter, the entire drum kit comes from the leakage from the toms. While this leakage may deliver a nice fat snare sound, the tom itself might not sound at its best when it's struck. A little dynamic EQ dip in the 300 to 400Hz range, or a stick enhancement at 5kHz can give you the full tom tone you're looking for without diminishing the strength of the snare.

Cymbals

Cymbals might sound great throughout a song until that loud section when the drummer gets carried away with a series of mighty whacks that stick out like a car crash. A dynamic EQ dip at between 5k to 10K will leave the cymbals' sound intact while taming the loudest hits.

Bass

The bass and kick might sound great until they're hit together and cover each other up. Assigning the kick to the sidechain input of a band of dynamic EQ set to 80 to 100Hz can clean up both instruments and keep them from clashing.

Mix

Sometimes a mix needs some high frequency "air" around 10kHz or above to open it up a bit, but those frequencies could make the mix too harsh when the cymbals really fire. Setting a dynamic EQ to dip slightly whenever the cymbals play will keep the air frequencies lively while taming the harshness.

There are many uses for a dynamic EQ that can be quite helpful during mixing. The key is to get as comfortable using dynamic EQ as you are with a static EQ and you'll find that your mix can benefit greatly from its inclusion.

Intelligent EQ Plugins

Equalizing used to be all about the mixer's ears but the latest generation of EQ plugins use a form of artificial intelligence that makes equalizing more about visually identifying a problem than having the aural experience to understand that a problem even exists. While less-experienced mixers can benefit greatly from these new processors, even veteran engineers now use them because of the speed they offer over the ears-only approach.

Most new equalizers incorporate a built-in frequency display. Sometimes, this display is merely for reference but in many cases it is an interactive tool to help find the frequencies that need correction. With some models, like the Fabfilter Pro-Q 3, you can actually grab a peak or trough on the display panel and insert an EQ band, which means you can essentially equalize on the fly.

EQ Match

Some equalizer plugins from companies like iZotope, Fabfilter and Waves, among others, have a feature known as EQ Match that allows you to match the tonal characteristics of a reference audio signal. If a vocal was recorded on a previous day or in another studio, and an overdub or fix pass sounds different, EQ Match can bring the tonal qualities closer together with minimum effort.

Usually, a piece of the previous audio program has to be analyzed by the processor before it can be matched. Also, it may take several tries until a suitable match occurs but the result is usually worth the effort.

Masking Meter

Some equalizer plugins, like the iZotope Neutron 3 or Fabfilter Pro-Q 3, can show a visual display of the frequency response of two DAW channels at the same time, highlighting the areas where the frequencies of the two overlap. This means that one channel will be masking those frequencies on the other, causing both mix elements to clash.

Neutron 3 (see Figure 9.7) actually links the equalizer bands of both channels together so when an area is boosted on one channel, it's cut on the other, providing better definition for both channels in the mix.

Figure 9.7: iZotope Neutron 3

Courtesy iZotope

Tonal Balance

Tonal Balance is a feature of Neutron 3 that shows the distribution of frequencies of a song. While all frequency analyzers will show this, the difference here is that you can compare your mix against a set of target track references and genre standards. From here, you can articulate the

frequencies of the individual track equalizers to better balance your track against its target frequency profile.

Self-Adjusting Frequency Bands

Oeksound Soothe 2 is a dynamic equalizer with self-adjusting frequency bands. Unlike traditional EQ-tools, Soothe 2 analyses the signal on the fly and adjusts the frequency based on the input (see Figure 9.8). This saves you from having to manually notch the problematic mid and high frequencies. The frequency band gain reduction kicks in when-and-where needed, without affecting the nearby frequency areas. The result is more detail, top end, and presence.

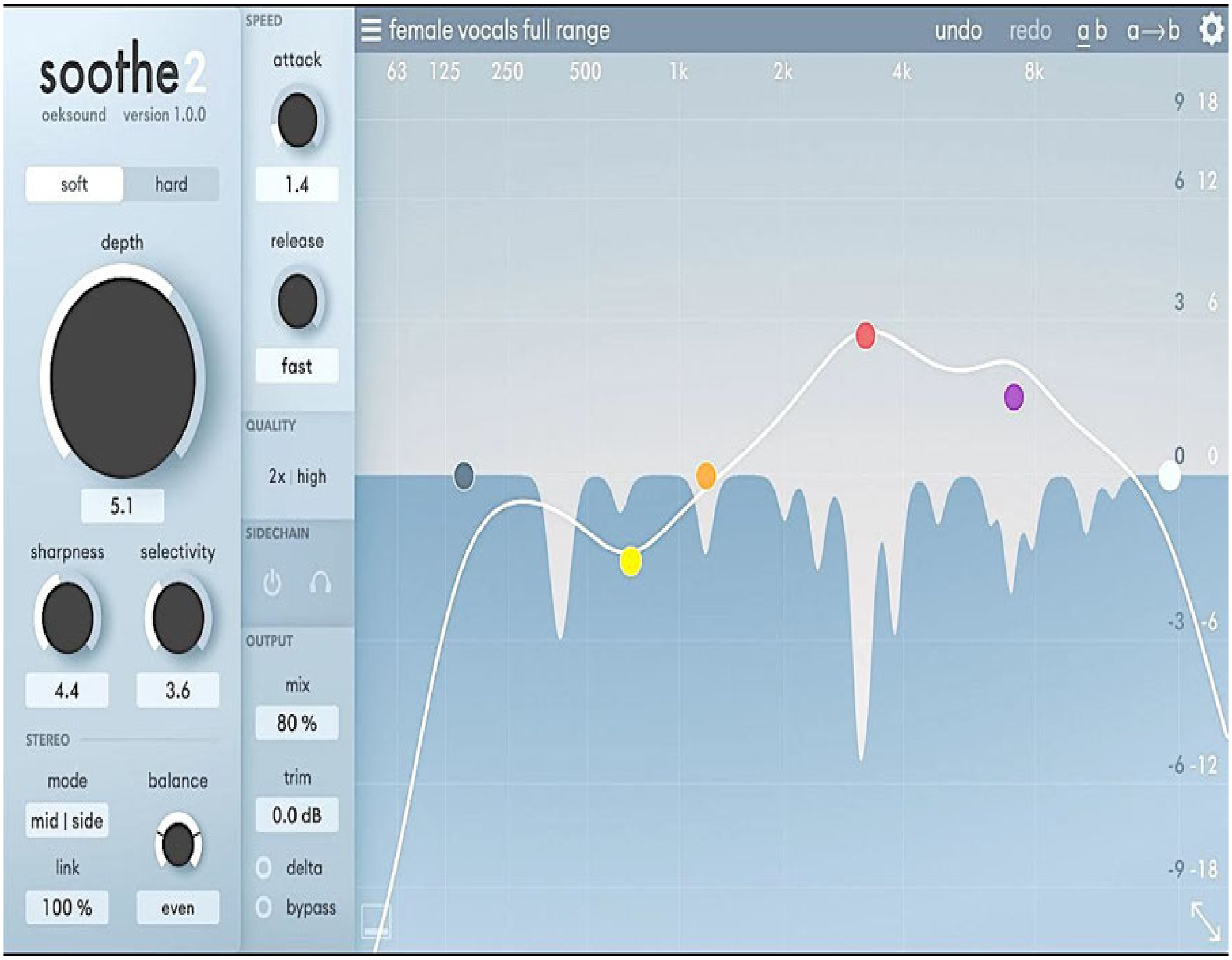

Figure 9.8: Oeksound Soothe 2

Courtesy Oeksound

Beyond Simple Equalization

Eventide introduced a new type of equalizer called SplitEQ that combines a parametric EQ with a transient designer (see Figure 9.9). This allows a mixer to apply equalization to just the transient part of the signal. With SplitEQ, it's easier to make mix elements jump out of a mix without them sounding like they've been EQed. This is especially useful

for fixing problems and shaping a mix that can't be easily achieved using other methods.

Figure 9.9: Eventide SplitEQ

Courtesy Eventide

The features above only scratch the surface of what the latest smart processors have to offer; with new previously unimaginable features

coming every season as more powerful, more intelligent processors are introduced. It’s a new and exciting territory providing the mixer with many more options and more precision in adjusting the frequency of a sound element than ever before.

Filters

A filter is a more limited form of equalization that only cuts or “rolls off” the frequencies below an assigned frequency point. These are useful for eliminating high or low frequency sounds that don’t add anything to a mix yet can have a negative effect on it, for example, truck traffic noise or wind.

Types Of Filters

There are multiple types of electronic and digital filters but the first two are mostly used in audio and most commonly found on channel strips and equalizers. They are:

High-Pass Filter

This filter type allows the highs to pass and the lows are cut. This occurs at a frequency point that is either fixed or variable. Sometimes called a “low-cut” filter.

Low-Pass Filter

The low-pass filter allows the lows to pass while the highs above a certain frequency are cut. Sometimes called a "high-cut" filter.

Notch Filter

This filter cuts a band of frequencies while allowing the frequencies on either side of the band to pass. Notch filters were often found on the old passive-filter sets by White Instruments and Altec that were used for tuning studios and venues back in the analog days. Notch filters aren't seen much today thanks to the advent of far more versatile (not to mention cheaper) digital instruments.

Bandpass Filter

Basically, the mirror image of a notch filter, a bandpass filter allows only a certain band of frequencies to pass while all the frequencies on either side of the band are cut. This type of filter isn't seen much in audio nowadays, although multi-band compressors use a similar idea.

Filter Paramenters

High and low pass filters don't have as many parameter controls as equalizers since boost and cut is not something a filter does by nature. Usually a Q (a term that describes a filter's quality, or Q, but really means bandwidth) selects how fast the filter will roll off, with 6dB per octave being the most gentle, 12dB per octave being the most usually selected, and 24dB per octave being the steepest; although some of the newer equalizers on the market also feature a "brickwall" filter, which cuts all frequencies off immediately below the frequency selection point. Until recently, brickwall-type filters were not included as they tended to degrade the sound quality but that's changed with some of the latest, more intelligent processors.

Frequency selection, or "cutoff", selects the frequency point where the filter will begin to take affect on the frequencies either above the limit (as in the case with a low-pass filter) or below it (a high-pass filter).

EQ Methods

Since each specific song, arrangement, instrument, and player is unique, it's impossible to give anything other than general guidelines when it comes to equalization techniques. That said, there are a number of methods that can quickly and easily get you in the ballpark, as long as you know what you're aiming for, musically. Remember that different engineers have different ways of arriving at the same end, so if a suggested method doesn't work for you, keep trying. The procedure doesn't matter, only the end result.

Before these methods are outlined, it's really important that you observe the following:

Listen! Open up your ears and listen carefully to all the nuances of the sound. Everything you hear is important.

Make sure you're monitoring at a comfortable level—not too loud and not too quiet. If it's too quiet, you may be fooled by the non-linearity of your speakers and overcompensate. If it's too loud, certain frequencies may be masked or overemphasized by the non-linearities of human hearing, and again you will overcompensate.

Method One: Equalize For Definition

Even source material that's been recorded well can lack energy due to certain frequencies being overemphasized or others being severely attenuated. More often than not, the lack of definition with a musical instrument can be blamed on too much lower midrange in approximately the 400 to 800Hz range. An imbalance in this area adds a "boxy" quality to the sound. Sometimes, a flat recording is because the sound is weak in the 3kHz to 6kHz area which makes it drab and undefined. Mixers hoping to zero-in on frequencies that mask definition in their tracks use a method called Subtractive Equalization. Here's how it works:

7.1. Set the Boost/Cut control to a moderate level of cut, 8 or 10dB should work well to start.

7.2. Sweep through the frequencies until you find the frequency where the sound has the least amount of boxiness and the most definition (see Figure 9.10),

or…

Sweep through the frequencies until you find the frequency that leaps out as offensive.

Adjust the amount of cut to taste, but be aware that too much cut makes the track sound thinner.

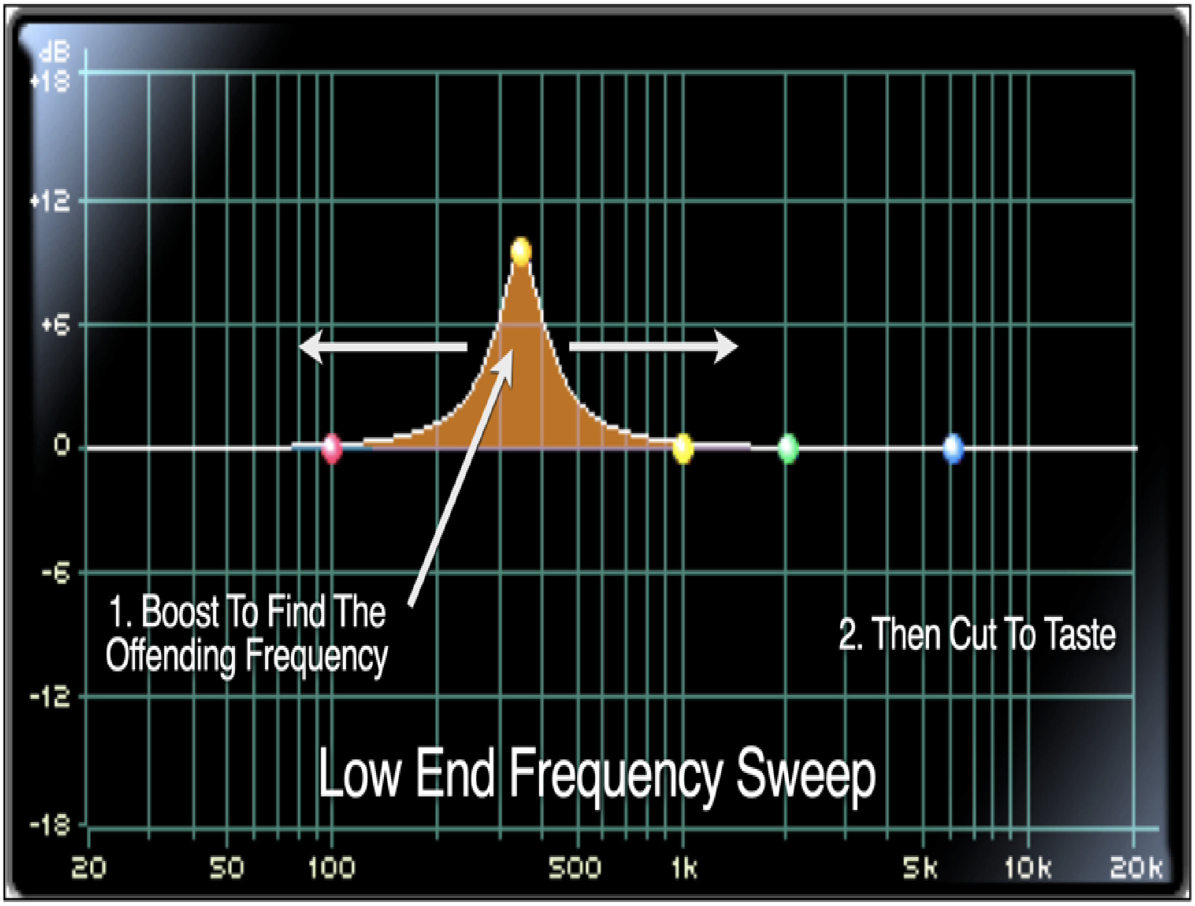

Figure 9.10: Low-end frequency sweep

There are two spots in the frequency spectrum where the subtractive equalization is particularly effective: between 200Hz and 600Hz and between 2kHz and 4kHz. This is because most directional microphones provide a natural boost between 200 to 600Hz thanks to the proximity effect brought about by close-miking; and many mics, especially those known for being good vocal mics, add a presence boost between 2kHz and 4kHz. Dipping those frequencies a few dB (more or less, as needed) can make a track sound much more natural than if you were to boost the other frequencies around it.

If there were a limited number of microphones (or even just one) used to record all the instruments in a home studio, these two frequency bands — or any band where there's a peak in the response — will tend to build up as more and more instruments are added to the mix. By dipping these bands a bit, you'll find that many instruments can sit better in the mix without having to add much EQ at all.

"What I hate to see is an engineer or producer start EQing before they've heard the sound source. To me, it's kinda like salting and peppering your food before you've tasted it. I always like to listen to the sound source first, whether it's recorded or live, and see how well it holds up without any EQ or whatever."

—Bruce Swedien

TIP: Always try attenuating (cutting) the frequency, as opposed to boosting it. This is preferable because all equalizers add phase shift as you boost, which can produce an undesirable coloring of the sound. Generally, the more EQ you add, the more phase shift is also added and the harder it may be to fit the instrument into the mix as a result. Many engineers are judicious in their use of EQ, but that being said, anything goes! If it sounds good, it is good.

Alternate Method, Equalize For Definition:

Starting with your EQ flat, remove all the bottom end below 100Hz by turning the low-frequency control to full cut.

Using the rest of your EQ, tune the mid-upper midrange until the sound is thick yet distinct.

Round it out with a supporting lower-mid tone to give it some body.

Slowly bring up the mud-inducing bottom end; raise it enough to move air, but not so much as to make the sound muddy.

Add some high-frequency EQ for extra definition (see Figure 9.11).

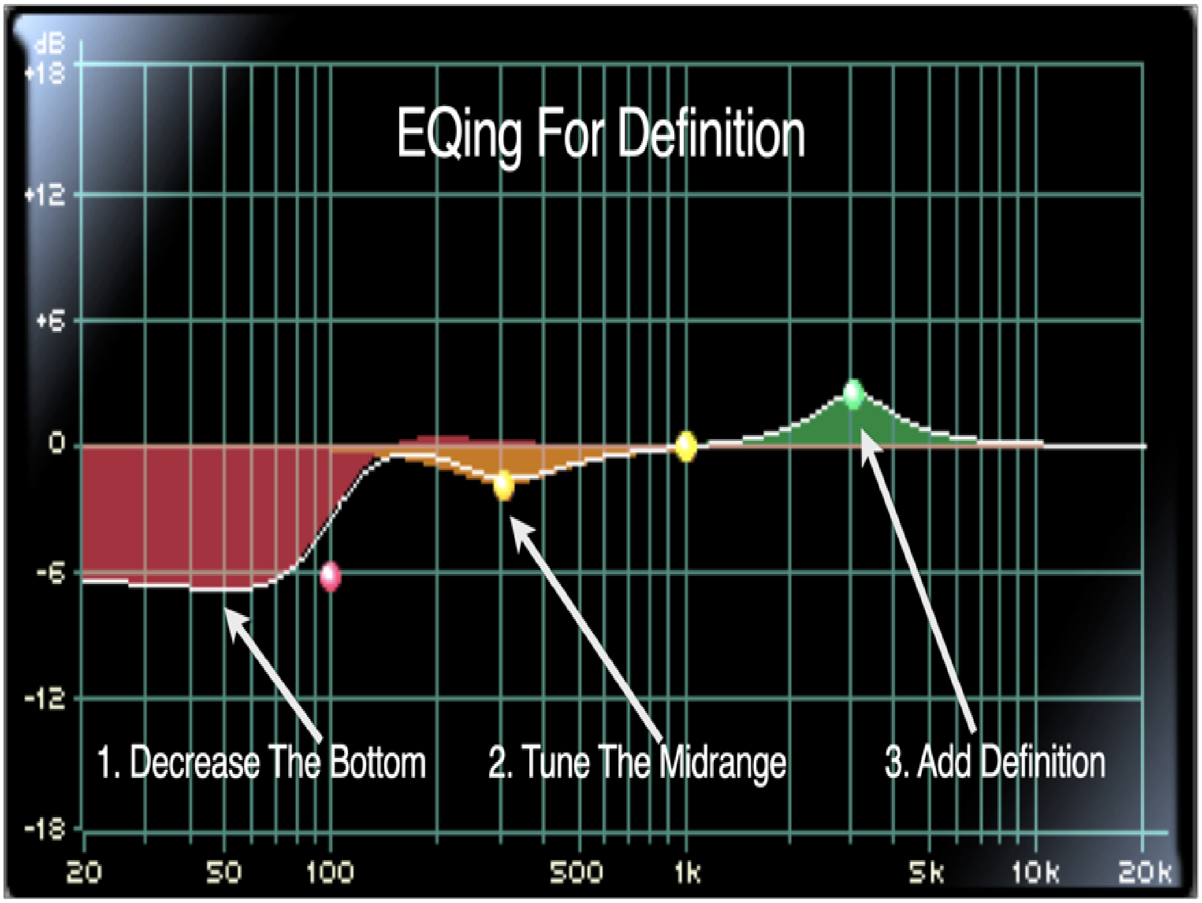

Figure 9.11: EQing for more definition

"I just try to get stuff to sound natural, but at the same time be very vivid. I break it down into roughly three areas: mids, the top and the bottom; then there's low mids and high mids. Generally, except for a very few instruments or a few microphones, cutting flat doesn't sound good to most people's ears, so I'll say, 'Well, if this is a state-of-the-art preamp and a great mic and it doesn't sound that great to me, why?' Well, the midrange is not quite vivid enough. Okay, we'll look at the 3k, 4k range, maybe 2500. Why don't we make it kind of come to life like a shot of cappuccino and open it up a little bit? Then maybe I'm not hearing the air around things,

so let's go up to 10k or 15k and just bump it up a little bit and see if we can kind of perk it up. Now, all that sounds good, but our bottom is kind of undefined. We don't have any meat down there. Well, let's sweep through and see what helps the low end. Sometimes, depending on different instruments, a hundred cycles can do wonders for some instruments. Sometimes you need to dip out at 400 cycles, because that's the area that sometimes just clouds up and takes the clarity away, but a lot of times adding a little 400 can fatten things up."

—Ed Seay

Method 2: Equalize For Size

Making a sound bigger or larger than life usually comes from the addition of bass and sub-bass frequencies in the 40Hz to 250Hz range, although most will come from an area just below 100Hz, a region just above 100Hz, or both.

To use this method, the low-frequency band of your EQ must be variable.

Set the Boost/Cut control to a moderate level of Boost (8 or 10dB should work).

Sweep through the frequencies in the bass band until you find the frequency where the sound has the desired amount of fullness.

Adjust the amount of Boost to taste. Be aware that too much Boost will make the sound muddy.

Go to the frequency either half or twice the frequency that you used in Step 2 and add an amount of that frequency as well. Example: If your frequency in Step 2 was 120Hz, go to 60Hz and add a dB or so as well. If your frequency was 50Hz, go to 100Hz and add a bit there (see Figure 9.12).

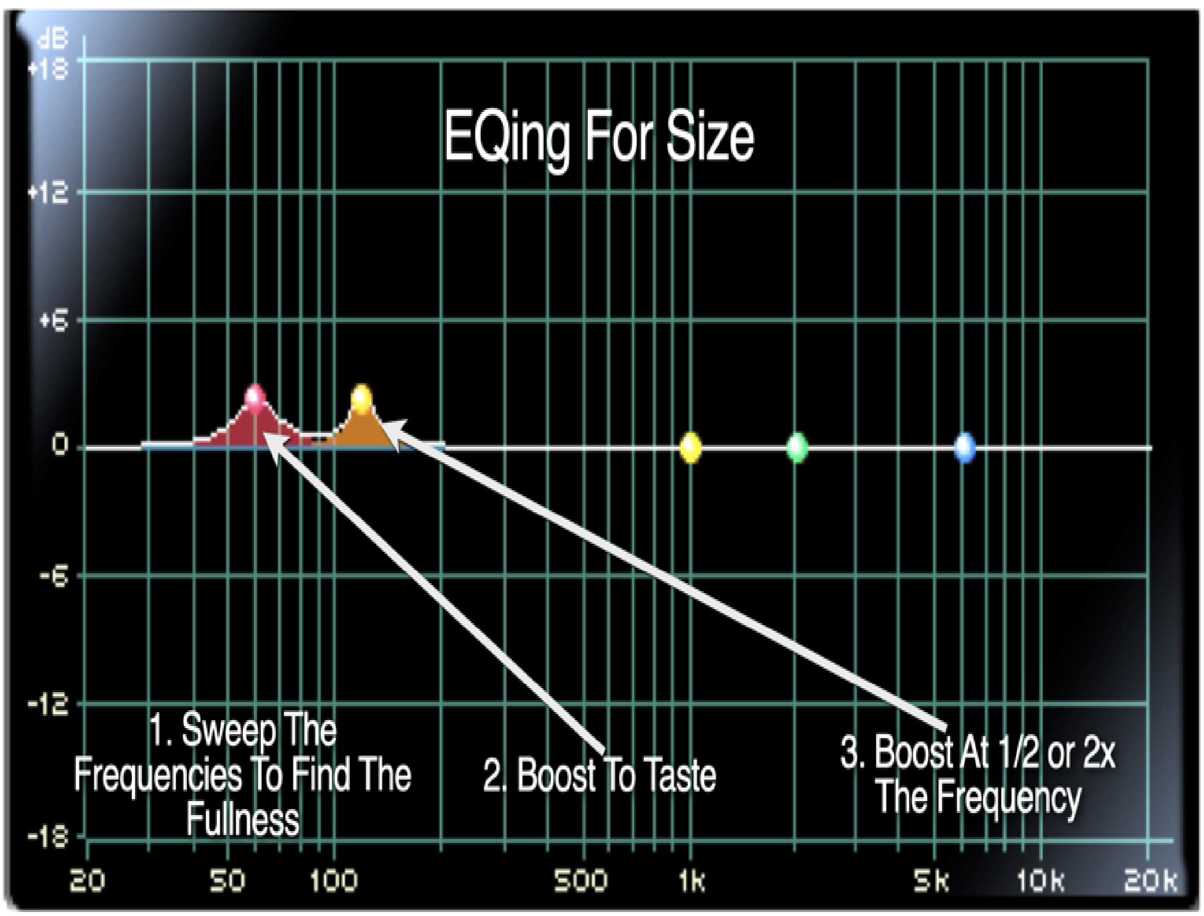

Figure 9.12: EQing for size

"I feel that equalizers are best used when used the least. I use them most to get rid of tones that are somehow not flattering. I'll most often use parametrics, sharp and subtractive, to look for the two or three biggest out-of-sorts characteristics. A snare drum for instance, has any number of boinks that I'll locate, and I may take them out or bring them up as I'm listening to the whole presentation, but I'll already know what and where they are."

—George Massenburg

Method 3: Juggling Frequencies

Most veteran engineers know that if you solo an instrument and equalize it without hearing the other instruments, you will probably end up chasing your own tail as you work to make each instrument bigger and brighter. When that happens, you'll quickly discover that the instrument you're EQing will begin to conflict with other instruments — or vocals — frequency-wise. That's why it's important to listen to other tracks while you're EQing. By juggling frequencies, they'll fit together better so that each instrument has its own predominant range. Here's how it's done.

Start with the rhythm section (bass and drums). The bass should be clear and distinct when played against the drums, especially the kick and snare. You should be able to hear each instrument distinctly. If not, do the following:

Make sure that no two equalizers are boosting at the same frequency. If they are, move one to a slightly higher or lower frequency.

If an instrument is cut at a specific frequency, boost the frequency of the other mix element to that same frequency. For example, if the kick is cut at 500Hz, boost the bass at 500Hz (see Figure 9.13).

Add the next most predominant element, usually the vocal, and proceed as above.

Add the rest of the elements into the mix one by one. As you add each instrument, check it against the previous elements, as above.

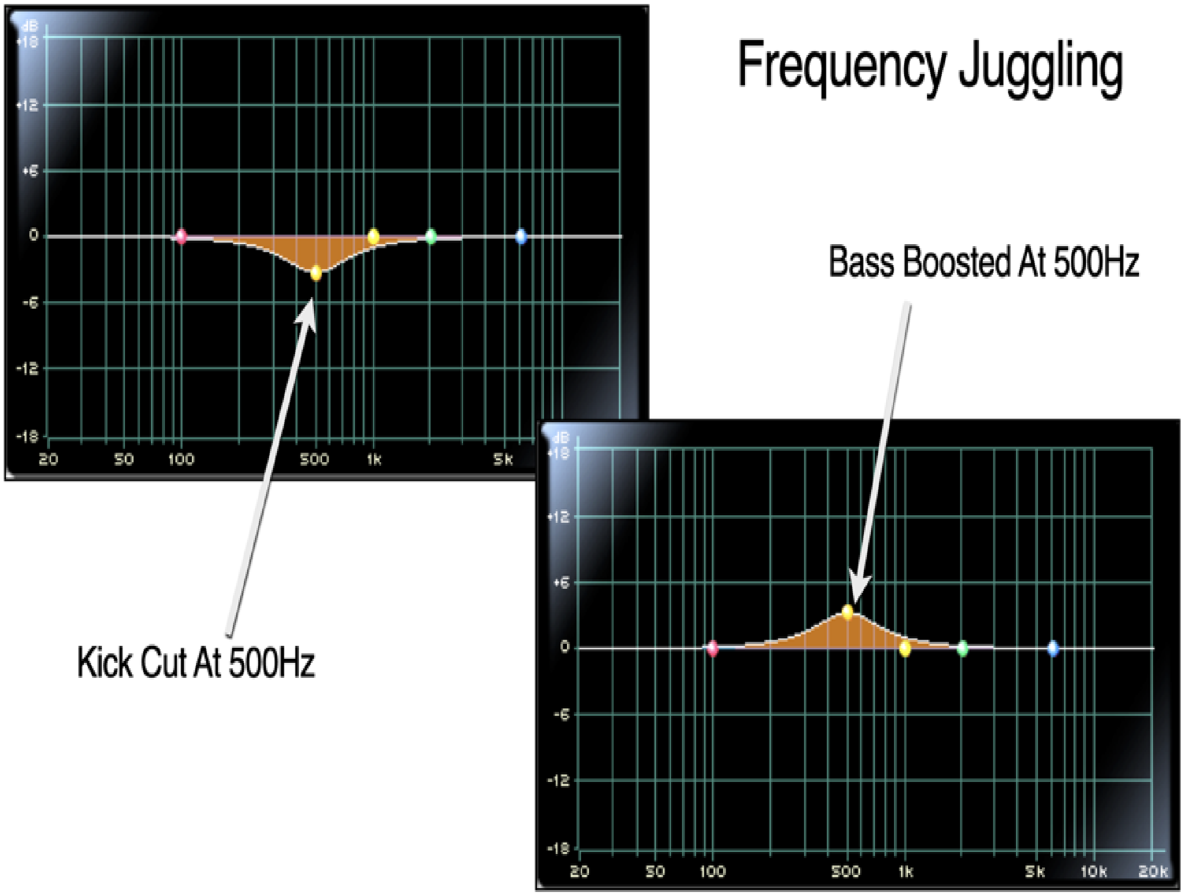

Figure 9.13: Juggling frequencies

The idea is to hear each instrument clearly and the best way for that to happen is to allow each instrument to nest in its own frequency.

TIP: It's not uncommon to EQ in a circle, where you start with one mix element, tweak another that's clashing with it, return to the original one to tweak it, and then go back again over and over until you achieve the desired separation.

“I really start searching out the frequencies that are clashing or rubbing against each other, but I really try to keep the whole picture in there most of the time as opposed to really isolating things too much. If there are two or three instruments that are clashing, that’s probably where I get more into the solo if I need to kind of hear the whole natural sound of the instrument.”

—Jon Gass

“Frequency juggling is important. You don’t EQ everything in the same place. You don’t EQ 3k on the vocal and the guitar and the bass and the synth and the piano, because then you have such a buildup there that you have a frequency war going on. Sometimes you can say, ‘Well, the piano doesn’t need 3k, so let’s go lower or let’s go higher,’ or ‘This vocal will pop through if we shine the light not in his nose, but maybe toward his forehead.’ In so doing, you can make things audible, and everybody can get some camera time.”

—Ed Seay

•You can't boost something that's not there in the first place.

Finding An Offending Frequency

Sometimes a sound has a frequency that sticks out of the mix like a pick-axe in your ear. The method to find it and attenuate it is similar to EQ Method One:

Set the Boost/Cut control to a moderate level of boost. Eight or 10dB should work. (see Figure 9.14)

Sweep through the band until the frequency that’s giving you trouble leaps out.

Adjust the amount of cut until the offending frequency is in balance with the rest of the sound. Be aware that too much cut can also decrease the definition of the sound.

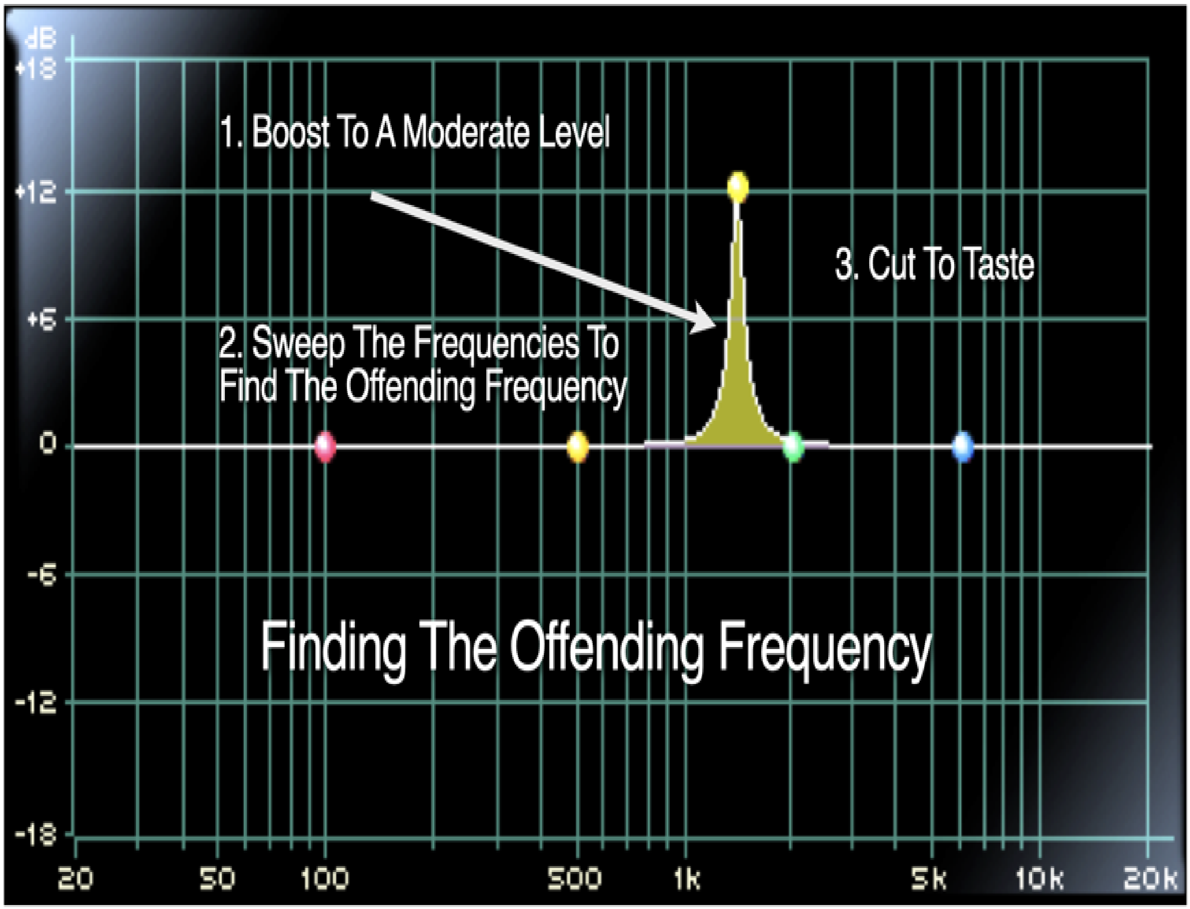

Figure 9.14: Finding the offending frequency

Method 4: Using Saturation To Color A Mix Element

Saturation is a relatively new category of signal processing that's being used increasingly to color the sound of a mix element in ways that weren't even considered until recently. Saturation processors like Softube Harmonics, Soundtoys Decapitator, SPL Vitalizer, Brainworx

bx_saturator and others can be used to help a track pop better in a mix without having to resort to equalization.

While a similar effect can sometimes be achieved using a distortion or amplifier-simulator plugin, Saturation differs in that it's a more subtle form of distortion; it adds more character and density than what we would normally classify as dirty or overloaded. Most saturation processors are modeled after classic analog hardware, and while some of these plugins are meant to be placed across the mix buss, many are now being used on individual instrument tracks as well.

Since there are no common controls between the different saturation processors, it's not feasible to provide a step-by-step method for their use, except for the following:

Insert the saturation processor on the desired track.

While listening to either the entire mix — or soloed against other tracks that may be masking it — set the processor's character controls until the track can be heard more distinctly.

Preventing Mix Element Clashing

One of the biggest problems with large sessions that have lots of sound elements is that, more often than not, a few tracks simply get lost in the mix. Much of this has to do with the fact that one track may be masking

another one, and this masking is a direct result of frequency bands that are clashing against each other.

Experienced mixing engineers know to find these clashing elements and minimize their effect but this effort is why a mix can take so much time to finish. One solution is to conduct a round-robin sequence of soloing two mix elements at a time and looking (listening) to see if one masks the other, then using the appropriate EQ to adjust the frequency balance so they can both be heard clearly; then repeating this sequence between two more tracks, and two more, and so on.

That said, there are a number of other tricks and tips that can be used to separate clashing tracks and make each stand out from the mix. Here's a simple checklist to help find your own best practices:

o Did you try muting one of the offending mix elements so that they both never play at the same time? Sometimes it's an arrangement issue that can be easily rectified by the mute button.

o Did you try lowering the level of the one of the offending mix elements? Sometimes one mix element is meant to support the other, not compete with it. Simply lowering the level of the clashing element might solve the problem.

o Did you try tailoring the EQ so that the one of the mix elements takes up a different frequency space? If both elements have a boost at 2kHz, for instance, the clashing element could be cut at that frequency so both can be heard. Or the clashing track can be boosted at 1kHz or 3kHz

instead (or another frequency area altogether) so they can both be heard separately in the mix.

o Did you try panning one of the offending mix elements? Panning the mix elements apart can be an effective way of eliminating the clash. Sometimes all that's needed is a few degrees of separation one way or the other. Be sure that nothing else is panned to that exact spot though, or you may create another clash.

o Did you try changing the arrangement and re-recording the track? Many times the problem isn't in the mix, it's in the arrangement where two similar sounding instruments are playing in the same register. Although an extreme measure, re-recording one of the parts with a different sound, a different part, or a different register most likely will clear up the problem.

TIP: Be aware that making a mix element sound great while soloed may create a clash with another mix element without you knowing it.

Mix element clashes are usually the result of poor arrangement and orchestration techniques, or piling on overdub band-aids in the hope of "sorting it all out in the mix". This can be done, but your mix will take a lot longer to complete as a result.

The Magic High-Pass Filter

One of the most useful and overlooked processors in a mixer's toolkit is the high-pass filter. The low frequencies of many instruments just get in each other's way and often don't add much to the overall sound. However, if you roll the low frequencies off below 100Hz (or even higher) on instruments other than the kick and bass, the mix begins to clean up almost magically (see Figure 9.15). Rolling off the low frequencies of a vocal mic can eliminate the rumble of trucks and machinery that you can't hear because they come in below the audible spectrum but still muddy up the mix. Sometimes, even rolling off the bass and drums at between 40Hz and 60Hz can make the mix a lot louder and punchier without noticeably affecting the low end.

Figure 9.15: The high-pass filter

TIP: After frequency juggling, an instrument might sound terrible when soloed by itself. That's okay. The idea is for it to work in the track, not in solo.

"It really doesn't matter what it sounds like by itself, because it has to work together with everything else. That's where some of the young producers blow it. They go through and solo tracks and make everything sound fat [by itself]; then when they put it all together, they have a big car wreck.

—Jon Gass

The Magic Frequencies

Every instrument has at least one frequency that might be considered "magic." With this in mind, you might want to try those frequencies first to make an instrument or voice sound fuller or more distinct. These Magic Frequencies are outlined in Table 9.4. But don't forget that the right frequencies are ultimately determined by how the mix element fits in the arrangement as well as other critical variables such as the studio's acoustics, the gear used, the signal path, and the musicians.

Table 9.4: Magic Frequencies for Mix Elements

Instrument	Magic Frequencies
Bass guitar	Girth at 50 to 80Hz, bottom at 120 to 2240Hz, attack at 700Hz, string noise at 2.5kHz
Kick drum	Bottom at 80 to 100Hz, hollowness at 400Hz, point/beater at 3k to 5kHz
Snare	Fatness at 120 to 200Hz, point at 900Hz, crispness at 5kHz, snap at 10kHz
Rack Toms	Bottom at 100 to 200Hz, fullness at 240 to 500Hz, attack at 5k to 7kHz
Floor Toms	Fullness at 80Hz, attack at 5kHz
Hi-hat	Roll off below 200Hz, clang at 1kHz, definition at 4kHz, sizzle at 8kHz
Cymbals	Clang at 200Hz, sparkle at 8k to 10kHz
Electric guitar	Fullness at 240 to 500Hz, presence at 1.5k to 2.5kHz, air/sizzle at 8kHz, attenuate at 1kHz for 4×12 cabinet sound
Acoustic guitar	Fullness at 80Hz, body at 240Hz, presence at 2k to 5kHz
Organ	Fullness at 80Hz, body at 240Hz, presence at 2k to 5kHz
Piano	Fullness at 80Hz, presence at 3k to 5kHz, honky tonk at 2.5kHz
Horns	Fullness at 120Hz, piercing at 5kHz

Vocal	Fullness at 120Hz, boomy at 240Hz, presence at 5kHz, sibilance at 4k to 7kHz, air at 10k to 15kHz
Strings	Fullness at 240Hz, shrill at 3kHz, scratchy at 7k to 10kHz
Conga	Ring at 200Hz, slap at 5kHz

6 Trouble Frequency Areas

Whenever an engineer has trouble dialing in the EQ on a track, chances are it's because of one or more of the Six Trouble Frequency Areas. These are areas where too much or too little can cause your track to either stick out like an alarm clock, or disappear into the mix completely. Let's take a look.

200Hz (Mud) - Too much here can cause the track or the mix to sound muddy or boomy, while not enough can make it sound thin. It's a fine line, but many times mixers err on the side of too much, ending up with a track that's too thick and clutters up the mix.

300 to 500Hz (Boxy) - Too much of this frequency area results in the dreaded "boxy" sound or, if you're listening to a floor tom or kick, the "beach ball" effect (where it sounds like a beach ball being hit). It's also a spot that some less-expensive microphones, especially dynamic mics, tend to emphasize which is why many mixers almost automatically cut a few dB in this range from the kick drum track during the mix.

800Hz (Walmart) - Too much in this area results in what's sometimes known as the "Walmart" sound, meaning that it sounds like a cheap stereo purchased in a department store. Try it for yourself: monitor through a cheap pair of computer speakers and you'll find that 800Hz is what you'll mostly hear. Obviously, too much of this frequency range is not a good thing.

1k to 1.5kHz (Nasal) - This is the nasal range of the frequency spectrum and, as the name suggests, too much results in a vocalist that sounds like they're singing through their nose. Once again, this is primarily a microphone problem, especially if it's poorly matched to the vocalist, but notching a bit out during the mix can fix it.

4kHz to 6kHz (Presence) - This frequency range is frequently underutilized during the mix, resulting in a track that lacks definition. Without presence, things tend to sound dull and leaden but too much of this sauce can make the track sound thin or reedy and, in the case of a vocal, sibilant.

10kHz+ (Air) - Another widely overlooked frequency range, the air band provides clarity and adds a certain "realness" to the track. Many vintage mics favor the air frequencies in their construction, which is why we prize them for their sound. The Maag Audio EQ4P has a special "Air Band" designed to provide those frequencies with a minimum of phase shift, but you can dial it in on other equalizers as well.

Sometimes, just a slight tweak to any of these six frequency ranges can elevate a mix from dull to exciting, or from muddy to clear; so keep them in mind during your next mix.

The Relationship Between Bass And Drums

Perhaps the most difficult task of a mixing engineer is balancing the bass and drums, especially the bass and kick. Nothing can make or break a mix faster than how these instruments work together. It's not

uncommon for a mixer to spend hours on this fundamental balance — both level and frequency — because if this relationship isn't correct, the song will never achieve the size and punch that modern ears expect.

So how do you find this mysterious balance?

The first rule is: you have to make space in your mix for both of these instruments. They need to coexist so they won't fight each other and turn the mix into giant puddle. While simply EQing your bass high and your kick low (or the other way around) might work at its simplest, it's best to have a more in-depth strategy to make them fit together. To that end, here's one approach that works extremely well:

EQ the kick drum between 60 and 120Hz, as this will allow it to be heard on smaller speakers. For more attack and beater click, add between 1kHz and 4kHz. You may also want to dip out some of the boxiness that lives between 200 and 600Hz, depending on the drum. EQing in the 30 to 60Hz range will produce a kick you can feel if your speakers are large enough, but can also make it sound thin on smaller speakers and probably won't translate well to a variety of speaker systems. Most 22-inch kick drums like to center somewhere around 80Hz.

Bring up the bass with the kick. The kick and bass should occupy slightly different frequency spaces. The kick will usually sit in the 60 to 80Hz range, whereas the bass will emphasize higher frequencies anywhere from 80 to 250 (although sometimes the two are reversed depending upon the song). Before you continue to EQ at other frequencies, try filtering out any unnecessary bass frequencies so the kick and bass are not boomy or muddy (below 30Hz on kick and 50Hz on the bass although, again, this varies according to style and taste).

There should be a driving, foundational quality to the combination of these two together.

A common mistake is to emphasize the kick with either too much level or too much EQ and not enough on the bass guitar. This gives you the illusion that your mix is bottom-light, because what you're doing is effectively shortening the duration of the low-frequency envelope in your mix. Since the kick tends to be more transitory than the bass guitar, this gives you the sense that the low-frequency content of your mix is inconsistent. For pop music, it's best to have the kick provide the percussive nature of the bottom while the bass fills out the sustain and the musical component.

Make sure the snare is strong, otherwise, the song will lose its drive when everything else is added in. This usually calls for at least some compression (see Chapter 9, "The Dynamics Element"). You may need a boost at 1kHz for attack, 120 to 240Hz for fullness, and 10kHz for snap. As you bring in the other drums and cymbals, you might want to dip a little of 1k on these to make room for the snare. Also, make sure that the toms aren't too boomy. (If so, try rolling them off a bit below 60Hz first before you begin to EQ elsewhere.)

If you're having trouble with the mix because it's sounding cloudy and muddy on the bottom end, mute the kick drum and bass to determine what else might be in the way in the low end. You might not realize that there are some frequencies in the mix that aren't musically necessary. With piano or guitar, you're mainly looking for the mids and top end to cut through, while any low end might just be getting in the way, so it's best to clear some of that out with a high-pass filter. When soloed, the instrument might sound too thin but combined with the rest of the mix the bass will sound so much better, and your ear won't be missing that low end from the other instruments. Also, the mix will sound louder, clearer, and fuller; but be careful not to cut too much low end from the other instruments as you don't want the final mix to lose its warmth.

For dance music, be aware of kick drum to bass melody dissonance. The bass line is very important and needs to work very well with the kick drum when it's reproduced over the monster sound systems commonly found in today's clubs. If your kick has a center frequency of an A note and the bass line is tuned to A#, they're going to clash. Tune your kick samples to the bass lines (or vice versa) where needed.

If you feel that you don't have enough bass or kick, boost the level, not the EQ. This is a mistake everyone makes when they're first working out their mixing chops. Most bass drums and bass guitars have plenty of low end and don't need much more, so be sure that their level is well balanced with each other and with the rest of the mix before adding EQ. Even then, a little bit goes a long way.

"I put the bass up first, almost like the foundation part, then the kick in combination with the bass to get the bottom. Sometimes you can have a really thin kick by itself, but when you put the bass with it, it seems to have enough bottom because the bass has more bottom end. I build the drums on top of that."

—Benny Faccone

"I'll get the drums happening to where they have some ambience, then put the vocal up and get that to where that's sitting right. Then I'll start with the bass and make sure that the kick and the bass are occupying their own territory and not fighting each other."

—Jerry Finn

Fitting It All In The Mix

One of the effective approaches to the equalization challenge is tailoring each mix element so it fits better in the mix. This is not only desirable, it is necessary. In the end, it's all about physics. There is only so much acoustic space in a mix so the more elements you try to squeeze in, the smaller sounding each one has to be.

Think of your mix like a bag and let's say this bag can hold four bricks. If we were to use smaller rocks we could fit in a lot more than four because they're all… smaller. The bag is only so big, however, so no matter how many rocks we want to fit into it, it will only hold so much.

In our final mix, we have to be able to fit in every sound frequency between 40Hz and 16kHz or so. If we're using only four mix elements, as in Figure 9.16, we can make each one bigger to fill up the total space. In practice, this means we can add more low end to each element and not overfill the capacity of the mix.

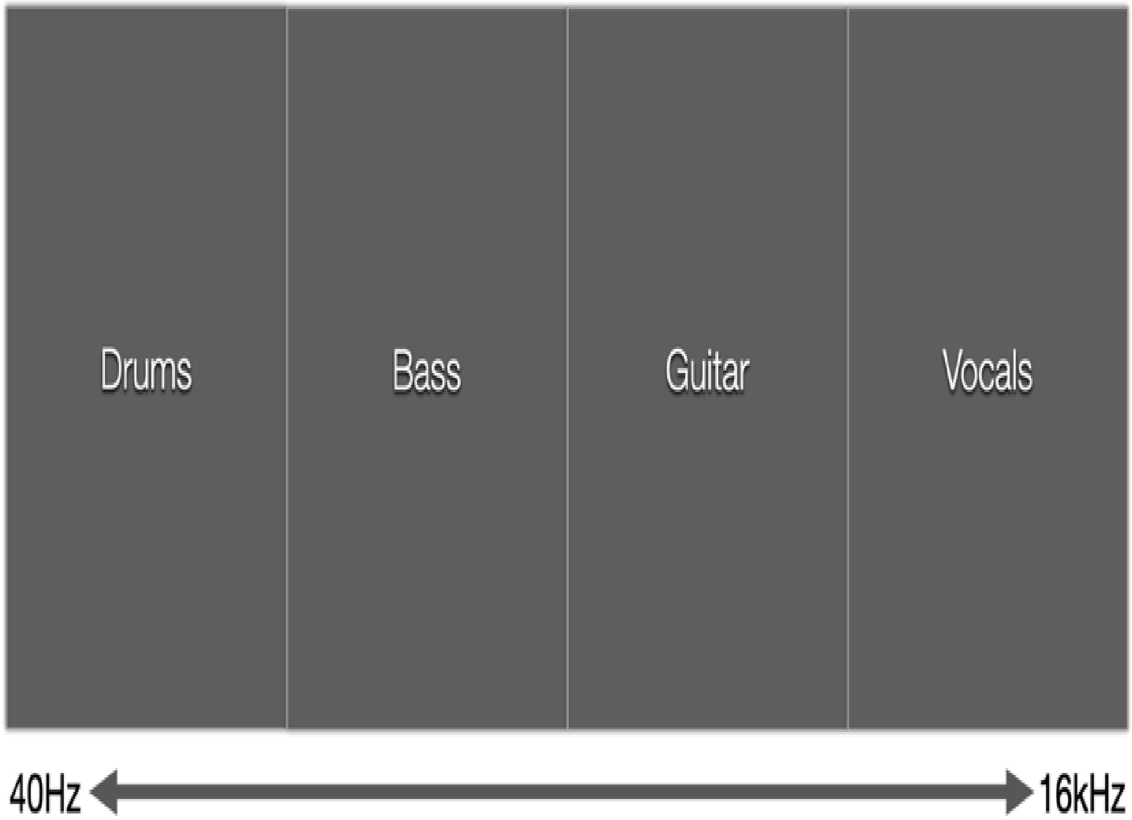

Figure 9.16: Each mix element bigger to fit in the mix

But in order to include more sound elements, as in, say, Figure 9.17, each element has to be smaller in order to fit in the same frequency range — the capacity of our bag, so to speak. That means we'll probably need to employ more high-pass filtering and EQing to make everything fit.

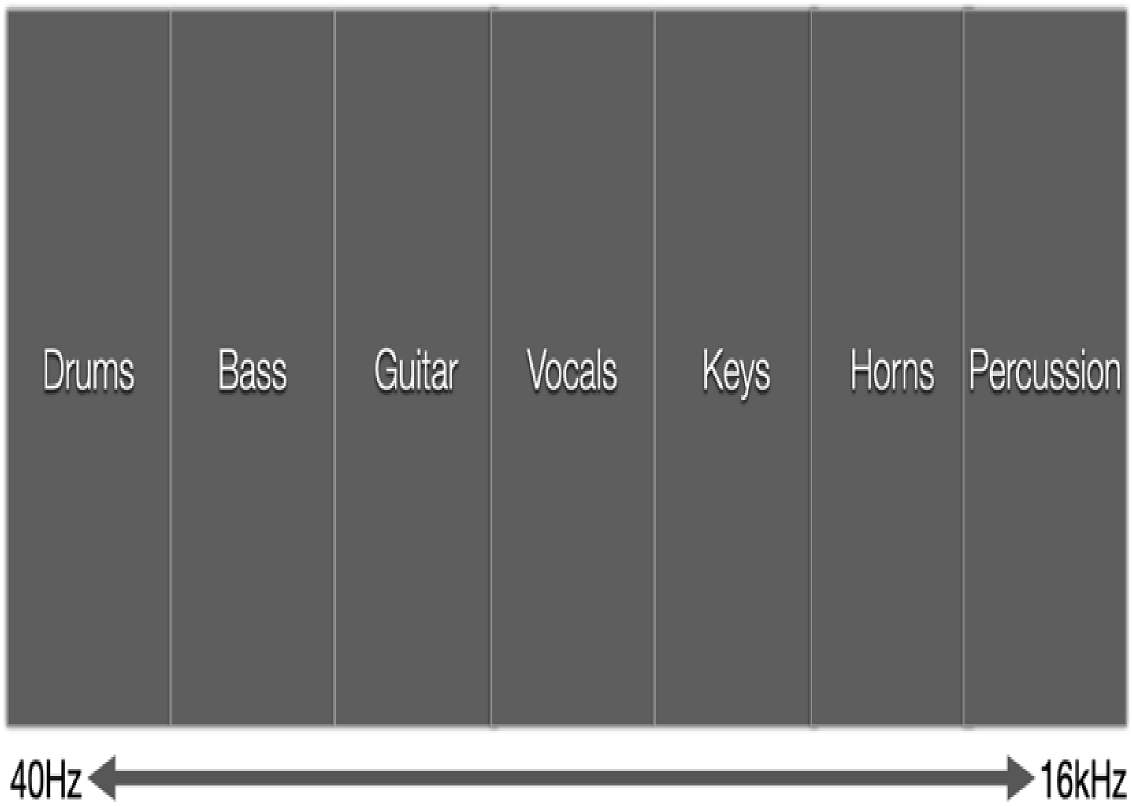

Figure 9.17: More mix elements means each one is smaller

EQ Techniques

Different mix elements require different approaches and, as we're beginning to understand, there's usually more than one method that will get the job done. Here are some tips — both general and specific — that have been used successfully in the past to produce world-class music:

General Tips

Use a narrow Q (bandwidth) when cutting. Use a wide Q when boosting.

If you want a mix element to stick out, roll off the bottom. If you want it to blend in, roll off the top.

The fewer mix elements that are in the mix, the bigger each one should sound.

Conversely, the more mix elements in the mix, the smaller each one needs to be in order to fit everything together.

It's usually better to add a small amount at two frequencies than a large amount at one.

Fine-tuning an instrument sound great while soloed may make it difficult (if not impossible) to fit together with the other instruments in the final mix.

For Snare

To find the "point" on the snare, boost the upper midrange about 5 or 6dB at 2kHz or so.

Open up the bandwidth (if that parameter is available) until you get the snare to jump out, then tighten the bandwidth until you get only that part of the snare you want most.

Now, fine-tune the signal until you need the least amount of boost to make the snare sit well in the mix yet still have its crisp definition.

For Drums

"In the old days you always pulled out a little 400 on the kick drum. You always added a little 3k and 6k to the toms. That just doesn't happen as much anymore because when I get the project, even with live bands. The producer's already triggered the sound he wanted off the live performance, so the drums are closer."

—Dave Pensado

For Kick

Try boosting 4kHz, cut 200 to 400Hz, and then boost the low frequencies around 80 to 100Hz to find the drum's resonance.

One fairly common effect used in R&B is to trigger a 32Hz tone with the kick, then gate it so it occupies the same volume envelope. Blend and compress both the original kick and the 32Hz tone to taste.

For a metal kick, add a bit of 3kHz or so on the kick drum to hear more of the beater.

For a kick meant to be played in a club, emphasize the 200 to 300Hz range while rolling off the extreme low end. The club system makes up the difference so if you mix the bottom of it the way you think you'll hear it in a club, you're probably going to overload the house amps.

If your bass delivers a very pure, sine wave-like sound and your kick is an 808, they may mask each other. If the kick is deeper than the bass, add a sample with some mid or top punch. If the kick is pitched higher than the bass, you can add some distortion or a plugin like MaxxBass (see Figure 9.18) to introduce a higher harmonic. Either way, make sure you check the results on small speakers.

Figure 9.18: Waves MaxxBass

Courtesy of Waves Audio Ltd.

For Bass

The ratio between the low bass (80 to 120Hz) and the mid bass (120 to 200Hz) is important. Try using two fairly narrow-peaking bands, one at 100Hz and another at 150Hz, boosting one and cutting the other slightly. If the bass is too "warm," sometimes reducing the lower band can make

it more distinct without removing the deeper fundamentals that live in the 100Hz band.

If you can't hear the bass when played on small consumer speakers, you're probably EQing too low. It's better to EQ the bass more above 100Hz than below that frequency.

For clarity on the bass, try boosting some of the 700-800Hz area, since this can provide definition without getting too "snappy".

A four-band parametric will allow you to adjust several bands below 200Hz. Try attenuating the low frequencies around 40 to 70Hz, then slightly boosting the frequencies from 80 to 120Hz where the fundamental lies, then boost the frequencies from 130 to 200Hz where the over- tones and cabinet/neck/body resonances live.

Muddy bass can mean a lot of things, but it usually involves a lack of presence of the higher harmonics. Most bass tracks have a sweet spot between 600Hz and 1.2kHz where the upper-order harmonics sing, so this is the place to boost for more presence in the mix.

Take a low-cut filter and center it at 250Hz so all the lows of the bass are attenuated. Now take a bell-shaped EQ and boost it 4dB with a narrow band and sweep around the 80 to 180Hz region to find where your bass frequencies fit best in the track. Once you find it, widen the bandwidth and boost more if necessary. If you want more density on the bottom, you may need to do this with another bell filter on the frequencies below the previous one. This should tighten up the low end, add space for a kick drum, and make your mix less boomy.

High-pass filter the bass anywhere from 40 to 80Hz. In many cases, it's amazing how often that one simple adjustment can help tighten up the bass.

Any mix elements with low-frequency content (below 500Hz) can affect the sound of the bass. This includes kick, keyboards, lower-register vocals, double bass, celli, low-tuned guitars, and so on. Cut the low end (anywhere below 80 to120Hz) from any mix elements where the low end isn't needed. This will help those tracks to cut through while leaving more space in the mix for the bass and the kick.

Removing signal information in the 250Hz region from mix elements such as guitars, keyboards, and even vocals often works better than cutting it from the bass.

With hip-hop and electronic music, the bass tends to contain a lot of information in the 30 to 60Hz range that you can literally feel. Many hip-hop or EDM records will raise the low-frequency target area slightly higher — to the 70 to 100Hz range — and elongate the duration to create the illusion that there's a lot of bass information and still sound deep and full on smaller monitors. Be careful not to over-EQ, though. Clubs (and cars!) with huge bass drivers are already pumping steroids in this frequency range.

With rock bass, the idea is to create an aggressive in-your-face energy so the focus here will be mainly on the amp output. Boost anywhere between 50 and100Hz for the bottom end, dip between 400 to 800Hz in the low-mids (this will allow the guitars and vocal to have more room to speak musically) and boost between 1.5 and 2.5kHz for midrange.

TIP: Be aware that mixing the DI sound with the amp sound might cause phasing problems in the midrange, so be sure to check the polarity with the phase parameter and use the selection with the most bass. Then again, sometimes the out-of-phase settings can work better in the mix, so be sure to check.

"The one thing that did change for me over time was my not liking 200Hz. That frequency couldn't be touched in the early days because we didn't have an EQ that was centered there, and it wasn't until later on that I decided that I didn't like it and began to pull it out [of the bass and kick]."

—Ken Scott

For Guitars

For a fatter-sounding guitar, boost the midrange about 9dB or so and sweep the frequencies until you find the range where the guitar sounds thick but still bright enough to cut through the mix. Then, back the boost down to about the point where the guitar cuts through the mix without being overpowering.

"I use EQ differently from some people. I don't just use it to brighten or fatten something up, I use it to make an instrument feel better. Like on a guitar, instead of just brightening up the high strings and adding mud to the low strings, I may look for a certain chord to hear more of the A string. If the D string is missing in a chord, I like to EQ and boost it way up to +8 or +10 and then just dial through the different frequencies until I hear

what they're doing to the guitar. I'm trying to make things more balanced in the way they lay with other instruments."

—Don Smith

Boosting 10kHz on guitars will accentuate finger noise and tiny string movements. Boosting 5 to 8kHz will allow the guitar to better cut through the mix. Boosting 1k to 5kHz will give the guitar a bigger presence.

Consider filtering the guitar with both high-pass and low-pass filters. If you leave too much low end in a distorted electric guitar, it will compete with the rhythm section. Too much above 8kHz can compete with the cymbals.

For Vocals

Give the 125 to 250Hz range a slight boost to accentuate the voice's foundation and make it more "chesty".

2k to 4kHz accentuates the consonants and brings the vocal closer to the listener.

"On a vocal sometimes I think, 'Does this vocal need a diet plan? Does he need to lose some flab down there? Or sometimes we need some weight on

this guy, so let’s add some 300 cycles and make him sound a little more important.’”

—Ed Seay

“I think of EQ as an effect, much the same way you would add chorus or reverb to a particular instrument or vocal. For example, I might have a vocal where I think it’s really EQed nicely, and then I’ll add a little more 3k just to get it to bite a little more. Then it just makes me feel like the singer was trying harder, and it brings out a little bit of passion in his or her voice.”

—Dave Pensado

10

The Dimension Element: Adding Effects

The fifth element of a mix is Dimension, which refers to controlling the ambient field that a mix element sits in. Dimension can be captured while recording but more commonly, it is created or intensified during the mix by introducing effects such as reverb, delay, or modulation. Adding Dimension might mean re-creating a complete acoustic environment but it could also mean enhancing the width or depth to a single track to define and improve an otherwise boring sound.

There are six reasons why a mixer might add dimension to a track:

To create an aural space

To add excitement

To make a track sound bigger, wider, or deeper

To move a track back in the soundfield (giving the impression it's farther away)

To add motion

To add a sheen or polish

"Everything has to be bigger always. Effects are makeup. It's cosmetic surgery. I can take a very great song by a very great band and mix it with no effects on it at all, and it'll sound good, and I can take the same song and mix it with effects and it'll sound fantastic!"

—Lee DeCarlo

"The way I think of it is the pan knob places you left to right while the effects tend to place you front to rear. In other words, if you want the singer to sound like she's standing behind the snare drum, leave the snare drum dry and wet down the singer, and it'll sound like the singer is standing that far behind the snare drum. If you want the singer in front of the snare drum, leave him dry and wet down the snare drum."

—Dave Pensado

"Sometimes [I add effects for] depth, and sometimes you just want it to sound a little bit more glamorous. I've done records where I didn't use any effects or any verb, but quite often just a little can make a difference. You don't even have to hear it, but you can sense it when it goes away. Obviously, an effect is an ear-catcher or something that can just kind of slap somebody and wake them up a little bit in case they're dozing off there."

—Ed Seay

One of the primary reasons we record elements in stereo is to capture the natural ambience (or dimension) of a mix element. Because we can't

always record everything this way, we frequently create this aural space artificially.

The Six Principles For Adding Effects

There are six principles that offer a general guideline on how effects can be used.

PRINCIPLE 1: Picture the performer in an acoustic space, then re-create that space around him or her.

This method usually saves time as it's a lot faster than simply experimenting with different effects presets until something excites you — although that method can work, too, but it generally takes longer. Keep in mind, the artificially created acoustic space you end up creating needn't be a natural one. In fact, as long as it fits the music, the more creative the better.

PRINCIPLE 2: Smaller reverbs or short delays make things sound bigger.

Reverbs with decays under a second (and usually much shorter than that) and delays under 100 milliseconds (again, usually a lot shorter than that) tend to make the track sound bigger as opposed to squeezing it back into the mix. This is especially true if the reverb or delay is in stereo.

Many times a reverb will be used with the decay parameter turned down as far as it will go (which may be as low as 0.1 seconds), but this setting is sometimes the most difficult for a digital reverb to reproduce, resulting in a metallic sound. If this occurs, sometimes lengthening the decay time a little or trying a different preset will result in a smoother, less tinny sound, or you can try another plugin or hardware unit that performs better under these conditions.

PRINCIPLE 3: Long delays, long reverb predelay, and reverb decay settings push a sound further away if the level of the effect is loud enough.

Delays and predelays (see the section later in the chapter on reverb) longer than 100 milliseconds are distinctly heard and begin to push the sound away from the listener.

The trick between something sounding big or just distant is the level of the effect. When the decay or delay is short and the level is loud, the track sounds big. When the decay or delay is long and loud, the track just sounds far away.

PRINCIPLE 4: If delays are timed to the tempo of the track, they add depth without being noticeable.

Most mixers set the delay time to the tempo of the track. (See the section on Delay for how to do this.) This technique makes the delay pulse in time with the music and adds a reverb ambience to the sound. It also

makes the delay seem to disappear as a discrete effect, but it still adds an overall smoothing quality or “glue” to the mix element.

PRINCIPLE 5: If delays are not timed to the tempo of the track, they stick out.

Sometimes you want to hear a delay distinctly and the best way to achieve this is to make sure the delay is not exactly timed to the track. Start by first putting the delay in time, then slowly adjust the timing until the desired separation is achieved.

PRINCIPLE 6: Reverbs sound smoother when timed to the tempo of the track.

Reverbs are timed to the track by triggering them off a snare hit and adjusting the decay parameter so the decay just dies out before the next snare hit. The idea is to make the decay “breathe” with the track. One way to accomplish this is to start by making everything as big as possible at the shortest setting, then gradually making the settings longer until it sits in time with the track. Predelay can also be timed to the track the same way as delays are.

Of course, the most important aspect of knowing when and how to add any effect comes down to one simple factor: experience. Nevertheless, keeping these principles in mind will give you a perfect place to start.

Using Delays

Delays are the secret weapon many mixers use when they want to to add depth to the mix without using reverb. If set up correctly, a delay can both pulse with a track and blend into it, making it feel both deeper and fatter without calling attention to itself.

Types of Delays

There are a number of different delays commonly used in a mix. Let's take a closer look:

Haas effect. This is a delay of 40 milliseconds or less that's not perceived as a separate repeat but can add a sense of spaciousness, especially if the delay is stereo or panned opposite the source.

Short. A short delay is generally anywhere from 40 milliseconds to around 150. The idea is to add a double-tracked effect to the sound. This can be heard on early Elvis records as well as others from that era and is sometimes known as a "slap-back" echo.

Medium. A medium delay is anywhere from 150 milliseconds to around 400. Even though we hear it as a distinct repeat, this length of delay is more for adding a sense of space around the source. It may be somewhat imperceptible if timed to the track.

Long. A long delay is anywhere from 400 milliseconds up to about a full second (1,000 milliseconds) or even more. You hear a long delay as a very distinct and specific repeat.

Stereo. A stereo delay allows for a different delay time on each side of the stereo soundfield.

Ping-pong. A delay that bounces from one side of the stereo image to the other.

Tape. In the analog days, delay was accomplished by using an outboard tape machine. The delay occurred because the playback head was located after the record head, which created a time delay (see Figure 8.1). As the speed of the tape machine changed, so would the delay. For example, a 15-IPS tape speed would result in a delay somewhere in the 125 to 175 milliseconds range (different models of tape machines offered different delay times because the gap between the heads was different for each). At 7½ IPS the delay would double to around 250 to 350 milliseconds. Because of the analog nature of magnetic tape, it has the characteristics of wow and flutter of the tape path, plus a rolled-off high-frequency response and increased distortion with each repeat, which most tape delay plugins try to emulate.

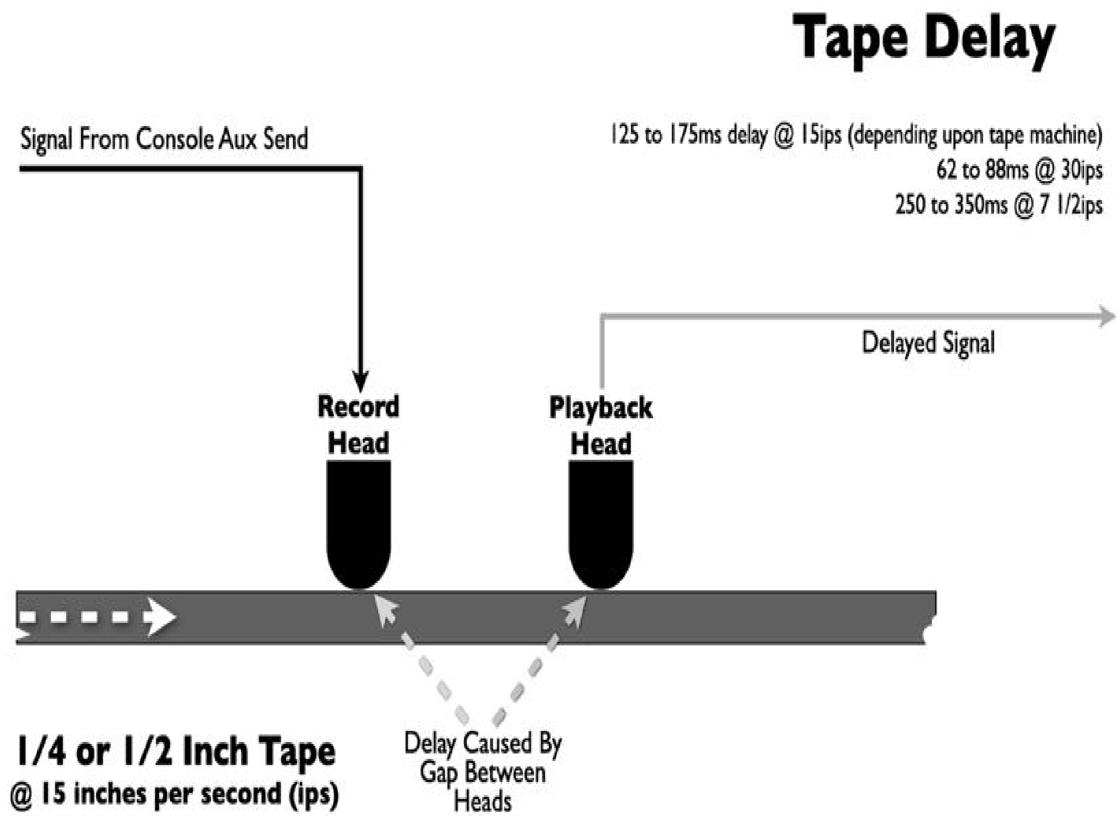

Figure 10.1: Delay using a tape machine

Timing Delays to the Track

It's important to time the delay to the track in order for it to pulse with the song. When this happens, the delay can almost seem to disappear into the mix, since you might not hear distinct repeats unless the track is soloed. Timing the delay to the track can add a depth or glue to the track that can't be achieved any other way.

Determining The Song’s Tempo

Before we can time the delay to a track, we first need to determine its tempo. Sometimes the tempo is predetermined in your DAW when the track is created since the beats per minute (bpm) is written into the session. Other times, many delay plugins allow you to tap it in. Alternatively, you can use an external tool like a smartphone app that shows the tempo of the track when you tap in it. That said, if you want to determine the bpm the old-fashioned way, here’s how it’s done:

The Classic Way Of Calculating The Tempo Of A Song

1. Count the number of beats in a song for 15 seconds.

2. Multiply the number of beats by 4.

For example: 24 beats × 4 = 96 bpm

Calculating The Delay Time

Delays are measured tempo-wise using musical notes in relation to the tempo of the track. For example, if the song has a tempo of 120 bpm, then the length of time it takes a quarter note to play would be 1/2 second (60 seconds ÷ 120 bpm = 0.5 seconds). Therefore, a quarter-note

delay would be 0.5 seconds or 500 milliseconds (0.5 × 1000ms per second), which is how almost all delay devices are calibrated.

But 500ms might be too long and using that delay setting might set the source track too far back in the mix. To calculate smaller delay increments, do the following:

Divide a 1/4-note delay in half for an 1/8-note delay (500ms ÷ 2 = 250ms).

Divide it in half again for a 1/16-note delay (250ms ÷ 2 = 125ms).

Divide it in half again for a 1/32-note delay (125 ÷ 2 = 62.5ms, or rounded up to 63).

If that's not short enough for the track, divide it in half again for 1/64 note (62.5 ÷ 2 = 31.25, rounded off to 31ms).

Again, this might not be short enough, so divide it in half again for 1/128 note (31ms ÷ 2 = 15.625, rounded up to 16ms).

And yet this still might not be short enough, so divide it again for 1/256 note (16ms ÷ 2 = 8ms).

Increments as small as 16ms or 8ms might not seem like much, but they're used all the time to make a mix element sound bigger and wider. Even a short delay like this will fit much more smoothly into the track if it's timed.

"I use a lot of 10, 12, 15ms on things. In the R&B stuff, you get a lot of stereo tracks that really aren't stereo. One of the first things I do is to widen the thing out, even if it's only 3, 5, or 10 milliseconds, and just get that stuff separated so I can keep my center cleared out."

—Jon Gass

It's also possible (and sometimes even preferable) to use other note denominations, such as triplets or dotted eights, sixteenths, and so on. These odd note denominations can be determined by using the following formula:

Delay Time × 1.5 = Dotted Value

Example: 500ms (quarter-note 120 bpm delay) × 1.5 = 750ms (dotted quarter note)

Delay Time × 0.667 = Triplet Value

Example: 500ms (quarter-note 120 bpm delay) × 0.667 = 333.5ms (quarter-note triplet)

And as with the straight notes (quarter, eighth, and so on), you can continually divide the above values in half until you get the desired denomination.

TIP: Triplet or dotted-note timed delays sometimes feel better in a track and give it more "glue" because of the way they blend with the pulse of the track.

Setting The Repeats

The number of repeats is selected by the Feedback control (sometimes called Regeneration), which sends some of the output of the delay back into the input. The amount is usually set in percentages with 0% resulting in a single repeat and 100% resulting in an infinite loop of repeats that eventually becomes feedback that gets louder and louder.

Although the number of repeats is dependent upon the song and its tempo, most of the time delays are set up for one to three repeats (less than 20% feedback). Too many repeats can sometimes result in the track sounding muddy.

The best way to set the repeats is to have only enough to either fill a hole in the arrangement or fill the space until the next event. In the case of a vocal, the repeats would occur from the end of a phrase only until the next phrase began.

Many delay plugins have a feature called dynamic delay, where the delay only begins after the source has died away and then automatically ceases the repeats when the next phrase begins again. This automatically keeps the delay out of the way, but the benefits of having a delay on all the time are negated.

Typical Delay Setups

Many mixers use standard delay setups of short and long; short, medium, and long; or even full note, triplet, and dotted note. Many times the delays will be set up during the session prep or via a mixing template, but the timing will be determined during the mix.

Here’s an example of a two-delay short and long setup, where if the tempo of the song is 105 bpm, the short delay might be set to 1/32 note of 71 milliseconds and the long delay might be set to a dotted 1/16 note of 214 milliseconds (see Figure 10.2).

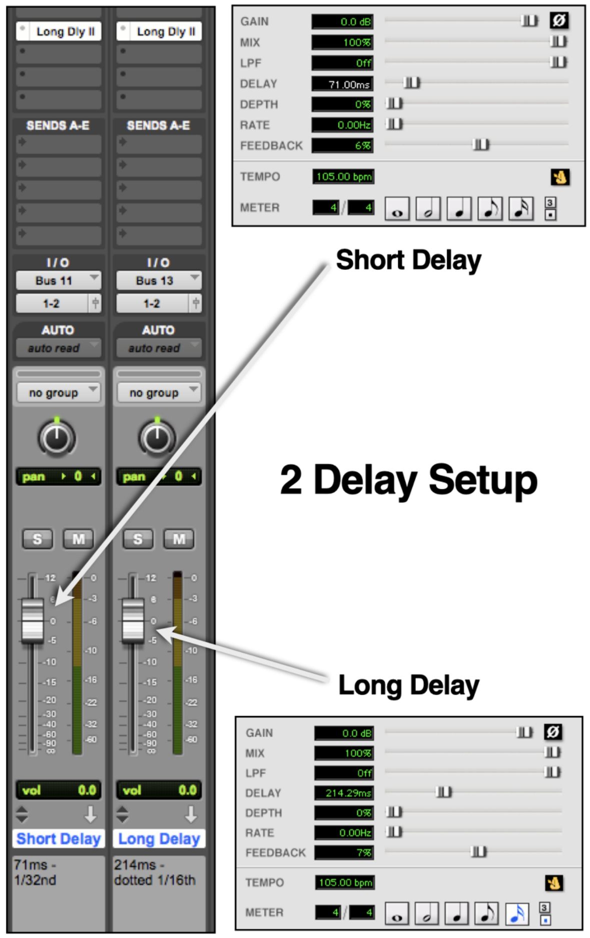

Figure 10.2: A two-delay short and long setup

In an example of a three-delay setup at a song tempo of 105 bpm, the short delay might be set to a 1/32-note triplet at 48 milliseconds, the medium delay to a 1/16 note at 143 milliseconds, and the long delay to a 1/4-note triplet at 381 milliseconds (see Figure 10.3).

Figure 10.3: A three-delay setup

Of course, these are only examples. In practice, note selections and delay timings are strictly based on how they feel in the song. It usually takes some experimentation to hear exactly what fits and what doesn't.

Delay Techniques

Here are a number of techniques often used for particular mix elements. Don't limit yourself to the examples cited though, as they can easily work for other instruments, vocals, or program sources as well.

For Vocals

A stereo delay with a ¼ or ⅛ note delay on one side and a ¼ or ⅛ note triplet or dotted note on the other provides movement along with depth and is a favorite trick of EDM mixers.

To simulate a vocal double, dial in a 1/16 note delay and then modulate it (see the "Using Modulation" section of this chapter) so it slowly raises and lowers in pitch. If the modulation can be set so it's random, it will sound more realistic.

For a quick vocal effect to give it some space and depth during tracking or overdubs, set up a mono 220ms delay with a couple of repeats.

Paul McCartney reportedly uses a 175ms delay on his vocals almost all the time.

For help getting a dry vocal to jump out, use a stereo bandwidth-limited delay (with a bandwidth of about 400Hz to 2.5kHz) set to the neighborhood of 12ms to the left and 14ms to the right, each side panned slightly off center. Bring up the delays until you can hear them in the mix and then back it off slightly. Occasionally mute the returns to

make sure it's still having an effect on the vocals out as all the channels sit with the rest of the mix. You can also time the delays to a 1/64 note on one side and a 1/128 note on the other.

"I like a vocal mostly dry, but then it usually doesn't sound big enough. You want the vocalist to sound like they're really powerful and dynamic and just giving it everything, so I'll put an 1/8-note delay on the vocal but subtract a 1/16, a 1/32, or a 1/64 note value from that 1/8 note. What it does is give a movement to the delay and makes the singer have an urgency that's kind of neat. I put the 1/8 minus 1/64 on the left side and put the straight 1/8 note on the right side. You can experiment with pushing the pitch up a little bit on one side and down on another too if your singer's a little pitchy, since that usually makes them sound a bit more in tune. Sometimes putting the ⅛ note triplet on one side and the straight 1/8 note on the other, if you've got any kind of swing elements of the track, will make the vocal big, yet it doesn't make the singer sound like he's taking a step back."

—Dave Pensado

For Guitars

To make the guitar sound larger than life, set a delay at less than 100ms (timed if you can) and pan the raw guitar to one side and the delay to the other.

During the '80s, when guitars were often recorded clean and direct, many LA session guitarists used a short stereo delay of 25ms on one side and 50ms on the other to provide additional space around the sound.

Use a mono delay on the guitar set to about 12ms (or whatever the tempo dictates) and pan the guitar and the delay to the sides. This makes the guitar sound much bigger and almost like two people playing perfectly in sync, yet still keeps a nice hole open in the middle for the vocals.

Pan the guitar track and the delay to the center (or put your monitors in mono), then slowly increase the delay time until it sounds bigger. Increase it a little more for good measure. You'll probably find the result is in the area of 25 to 30ms.

TIP: Instead of always syncing your delay to the tempo of the song in your DAW, try tapping the tempo in manually instead or set the delay slightly ahead or behind the beat for a little more natural groove.

For Keyboards

A stereo delay setting of 211ms on one side and 222 on the other provides a quick and easy room simulation and adds some life to a directly recorded keyboard.

Using Reverb

Reverb is one of the two principle ways we can add artificial ambience to a track in a mix, but there's a lot more to doing it properly than just dialing up a preset. Sure, that may work sometimes but more often than not, a reverb's parameter settings are as carefully crafted as the mix itself.

Types of Reverb

There are six primary categories of reverb, all with a different characters. Three of these are actual acoustic spaces, two are analog ways to reproduce the sound of a real space, and one is not found in nature but can really sound cool. The reason why there's a difference is that just like everything else in music and audio, there are many paths to the same result. You'll find that every digital reverb plugin or hardware unit produces its own unique version of these sounds.

Hall. A hall is a large space that has a long decay time and lots of reflections. Sometimes there's a subcategory of the hall reverb called "church," which is just a more reflective hall with a longer decay.

Room. A room is a much smaller space that can be dead or reflective, depending upon the material that the walls, floor, and ceiling are made of. It usually has a short decay time of about 1.5 seconds or less.

Chamber. An acoustic chamber is a dedicated, tiled room that many large studios used to build to create reverb (see Figure 10.4). For instance, 1960s hit-maker/producer Phil Spector's "Wall of Sound" was built around an excellent acoustic chamber at Gold Star Studios in Hollywood (long-since

closed, unfortunately). The acoustic chambers at Capitol Studios, designed by the legend Les Paul himself, still have a reverb sound that is revered by mixers everywhere (you can now get this on your DAW with UAD's Capitol Chambers plugin). Other artificial spaces commonly used as acoustic chambers include showers and stairwells.

Figure 10.4: An acoustic chamber

Plate. A plate is a 4 foot by 6 foot hanging piece of sheet metal with transducers attached to it that many studios used for artificial reverb when they couldn't afford or have the space to build a full-on chamber (see Figure 10.5). The first plate reverb was the EMT 140, developed in the late 1950s, that's still held in high regard by many mixers for its smooth, round sound.

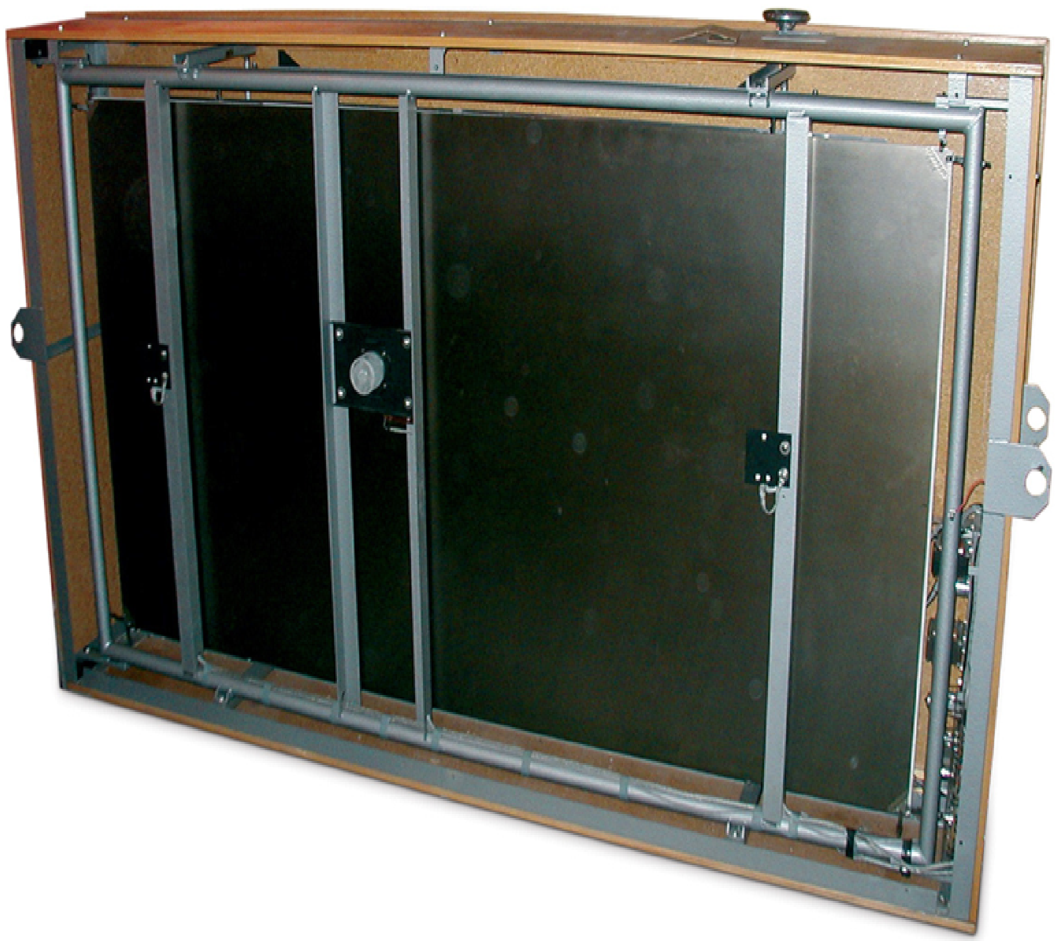

Figure 10.5: A plate reverb

Spring. A spring reverb is just that - a number of springs of different lengths that delay the audio signal at different times to create reverberation. Spring reverbs have always been an inexpensive, although limited way to create the effect. Some where inexpensive enough to be included on early Fender guitar amps of the 1960s, which was a big innovation at the time. Spring reverbs are generally a lot less flexible in terms of parameter control, although some of the larger units intended for studio use, including the AKG BX15(Figure 10.6) provided more control than most of the less expensive versions. Today, we have excellent software versions of spring reverb units available by numerous manufacturers, such as the PSP Springbox, that add far more parameters and control flexibility than their hardware counterparts.

Figure 10.6: A spring reverb

Non-linear. The non-linear category is strictly a product of modern digital reverbs, as the sound isn't found anywhere in nature. While natural reverbs decay in a rather smooth manner, when a reverb is created digitally, it's possible to adjust the decay in unusual, and virtually impossible ways. For example, the reverb tail can be reversed so it builds instead of decays, or it can be made to decay abruptly, both of which makes the decay "non-linear." This preset was a popular mixing effect used on drums during the 1980s when the feature first became available on the AMS RMX 16 digital reverb (see Figure 10.7).

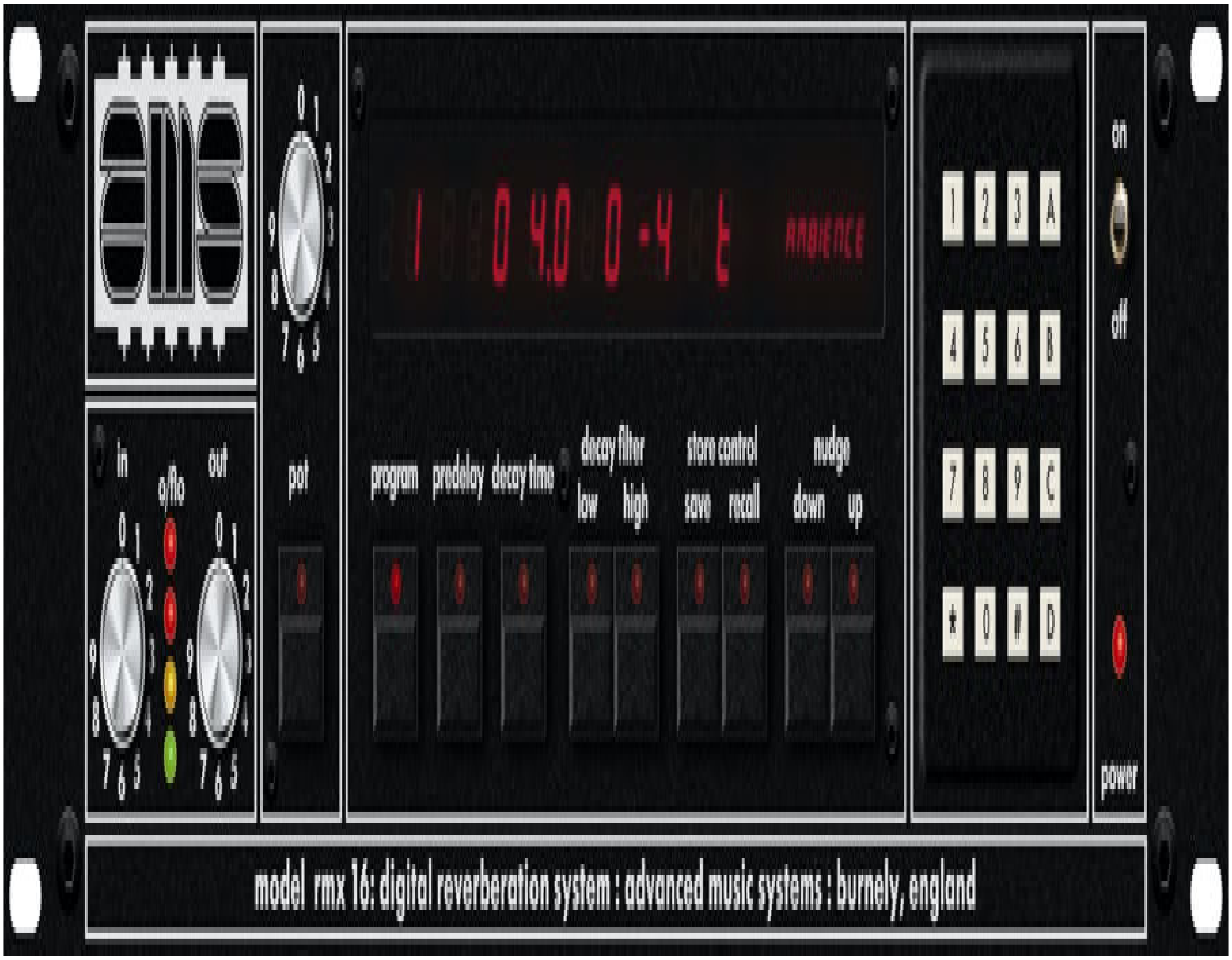

Figure 10.7: The AMS RMX 16 digital reverb

When it comes to deciding which reverb category to use, that's strictly up to the taste of the mixer; but there are some general guidelines.

Mix elements with sharp transients like drums and percussion may pair well with a reverb that has a naturally short decay time like a room, although a short decay plate or chamber can work as well.

Mix elements with sharp transients like drums and percussion also pair well with a non-linear reverb with a quick decay such as a reverb with a gated or non-linear setting.

Mix elements with long sustaining notes like strings and instrument pads pair well with halls or long decay plates and chambers.

Be aware that while many mixers might always use a room or a chamber on drums, a plate on vocals or guitars, and a hall on strings or keyboards, others may do just the opposite. Many might refine the sounds of each and finally settle on a few that they feel always work in a particular situation with a certain instrument or vocal. It's all comes with experience.

All of these reverbs can be modeled using what's known as a convolution reverb that uses a quick burst of audio energy (called an impulse) to excite the room or device, which then allows it to sample its parameters. Examples of convolution reverbs include the Audio Ease Altiverb and Avid's Space.

Timing Reverbs To The Track

As with delays, reverbs sound smoother if they're timed to the pulse of the track. Doing this adds depth without the reverb sticking out and makes the mix feel more polished. The two parameters that are adjusted for timing are the decay time and the predelay.

Timing The Decay

In simple terms, the decay is the time it takes for the reverb tail to die out. If the decay is timed to the pulse or bpm of the song, the track seems tighter and cleaner while still retaining all its depth.

To time the decay to the track, trigger the reverb with the snare hit and adjust the decay parameter so the decay just dies by either the next snare hit or the one after that. The idea is to make the decay “breathe” with the track. You can try to use this decay time to set other reverbs being used, but you’ll probably have to adjust them slightly because the decay response of every reverb or reverb algorithm (such as hall, plate, chamber, room) is different due the unique characteristics of the individual reverb.

TIP: If the decay sounds too short after just one hit, time it so the decay dies by the end of the second or even the fourth snare hit.

Timing The Predelay

Predelay means delaying the reverb entrance slightly after you hear the source signal. The reason it’s used is so the source signal doesn’t sound washed out in ambience. With a little bit of predelay, you’ll hear the source’s attack, then the reverb, so the source signal has more definition as a result.

Predelay is usually timed to the tempo of the track. Back in the days of physical plates and chambers, predelay was achieved using the slap delay from a tape machine (see Tape Delay described previously in the chapter), but today it's a standard parameter on every reverb plugin or hardware device.

The same way that you determined the delay time for the track provides the timing for the predelay. The difference is that you usually need a smaller increment to what you've already used for a delay, and it may be less than 100 milliseconds.

For instance, if you determined that a suitable 1/16-note delay time is 150ms, cut it in half (75ms), then cut it in half again (37.5ms), and maybe even in half again (19ms, rounding it off). That's probably going to be the best timing to start with, but don't be afraid to try the longer or shorter variations as well. That said, many mixers like to use a predelay of well over 100ms if the song calls for it.

TIP: A predelay in the 20 to 40ms range is the most common. If you don't want to time it, just start with 20ms as a good compromise.

Of these two parameters, the predelay is probably the most important in that the reverb seems more a part of the track when that parameter is timed. If you really want the reverb to stick out a bit, just randomly select a predelay time or use none at all.

Reverb Diffusion

As you've seen, most reverb plugins concentrate on the three parameters that count the most; the type (hall, room, plate, etc), the predelay, and the decay. If you really want to get tweaky with your effect though, there's another parameter that can be just as important, and that's reverb diffusion.

Natural reverb is made up of thousands of reflections off the surfaces of the environment that you're in. These are basically individual tiny echos, but they're strung together so closely that we don't hear each of them separately (remember the Haas effect where you can't hear individual echos as separate events below 40 milliseconds?).

Digital reverb duplicates these reflections by creating thousands of random echos, but one of the cool things is that because it's all in the digital domain, we can manipulate these echos to emulate different environments.

Reverb diffusion controls how many of those echos are generated and how they're separated from each other. Why is that important? Different surfaces affect the diffusion differently.

For instance, a surface that's smooth — like a gym floor or a reverb chamber with tiled walls — will have a low diffusion, which means there's a lower number of reflections that are more evenly spaced. An environment with a rocky surface and lots of jagged edges (like a cave) will have a high rate of diffusion, meaning there are a lot more reflections that are more randomly spaced.

Using The Diffusion Parameter

When it comes to real world use, drums and percussion tend to sound better with high diffusion since the random reflections tend to emphasize the transients.

On the other hand, long sustaining pads like strings and synths, or even vocals and horns, tend to sound better with low diffusion where the reflections are evenly spaced so the reverb is smoother and not as prominent.

There's another reverb parameter that's sometimes available called Density that's similar to Diffusion and produces similar results. Density determines how closely the reflections are to each other. The closer they are to each other, the smoother the reverb sound, especially the tail.

In the end, these two parameters aren't something that you need tweak on every mix. They're there for when you have a sound that's pretty much what you're looking for, but not quite there yet. Diffusion can sometimes dial in the perfect reverb for the situation.

Typical Reverb Setups

Just as with delays, many mixers use different combinations of reverbs. While some old-school engineers might only use a single reverb on a mix and it will sound great, some will, as a matter of course, use more.

In many cases, two reverbs are used with one set to short decay and used on drums and the second with a longer decay and used on the other mix elements. Figure 10.8 shows a typical two-reverb setup with parameters that will be timed to the tempo of the song (which, as with the example for the delays, is 105 bpm). The short reverb is set to a room reverb with an 18ms predelay (1/128 note) and a 1.2-second decay, while the long reverb is set to a plate sound with a 36ms predelay and a 1.8-second decay time.

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Figure 10.8: A two-reverb setup

TIP: The decay time can never accurately be predicted by the bpm of the song since the rate of decay is different for each reverb algorithm.

In a three-reverb setup, we may use different reverb categories but the decay is usually relegated to short, medium, and long. In Figure 10.9 the short reverb is set to a Plate with a decay time of 0.8 seconds, the medium reverb is a Chamber set to a decay time of 1.4 seconds, and the long reverb is a Hall set to a decay time of 2.2 seconds. If the bpm of the song is still at 105, we might start with all three predelays set to 18 milliseconds and fine-tune them from there as the mix progresses.

Figure 10.9: A three-reverb setup

TIP: Even when using multiple reverbs, one of them (usually the longest) may be used as a sort of unifying glue to make all the mix elements feel as if they're being performed in the same environment at the same time.

Reverb Techniques

Here are a number of techniques often used when adding reverb to particular mix element. Don't limit yourself to just these examples, as the settings can just as easily work for other mix elements in a variety of situations.

For Vocals

Automate the delay or reverb return so that during the sparse parts of the arrangement — particularly in the beginning of the song — the vocal is less wet and more upfront and intimate, which also makes the effect less obvious.

Try mixing various reverbs together. Set up three reverbs - short, medium, and long (the specific durations will vary with the song). On a non-ballad vocal, favor the short and medium over the long. The short one (try a 0.3 to 0.6 second, room or plate) will thicken the sound. Blending in the medium (1.2 to 1.6 second, plate or hall) will create a smooth transition that is quite dense but still decays fairly fast. Add a little of the longer reverb (2 to 3 second, hall) for whatever degree of additional decay you want to introduce. Combined, these three effects will sound like one thick reverb that will stick to the vocal and not muddy it up with excess length and diffusion.

With a singer/acoustic guitar player, try to picture the performer in an acoustic space and then realistically re-create that space around him or her. This lends itself to a medium-sized room or a small plate, with perhaps a little more reverb on the voice than the guitar. If the vocal is wet and the guitar dry (forgetting about leakage for a moment), it's difficult to have them both appear to share a common acoustic space.

If a vocal effect is too prominent, bring up the reverb to where you can hear it, then back off the level 2dB. Add a dB or two at 800Hz to 1kHz to either the send or return of the reverb to bring out the effect without it being too prominent.

For an interesting reverse reverb effect on a vocal where it's whooshing in before the vocal begins, set a reverb to a very long decay time (more than 4 seconds) and then record the reverb only onto a second track. Reverse it and move it forward on the timeline so it begins before the vocal.

For Drums

For the Tommy Lee "Thunder Drums" effect, set a reverb on the "cathedral" or "large hall" setting and then add a little to each drum. Pan the reverb returns so the reverb sits behind each part of the kit. For this effect to work, the bass drum has to sound tight to begin with and have a lot of beater present, and all the drums should be gated with the gate timed to the track. (See Chapter 8 for more on gates.)

For an "exploding snare" effect, add a short slap from 50 to 125ms with a touch of feedback to the bottom snare mic if there is one. Bring the slap back on a second channel. Using an aux send, send signal from both top and bottom snare mics and the slap to a short reverb of a second or less (timed to the song). By adjusting the proportions, phase, and EQ, the effect will fit it into almost any situation.

For Percussion

For hand percussion, such as shakers and tambourines, use a medium decay (0.8 to 1.2 seconds) room or plate reverb with either zero or very short (20ms) predelay.

For Guitars

To make guitars bigger, take a mono reverb and lower the decay time as low as it will go (0.1 seconds if it will go that low). Pan the guitar to one side and the reverb to the other. Try different reverb types to see which works better in the song. Increase the predelay and decay times slightly to make the sound bigger or to eliminate any metallic-sounding artifacts from the reverb.

For that early Eddie Van Halen sound, use either a chamber or a plate reverb set to about 2 seconds decay time and around 120ms predelay that's timed to the track. Pan the dry guitar to one side and the reverb to the other.

For Keyboards

For a keyboard pad sound that melts into the track, use a hall reverb with a 2 to 2.5 second decay and a medium to long (60 to 120ms) predelay that's timed to the track. Set any EQ or filters so that the extreme high and low ends are rolled off to about 8kHz and 150Hz.

For Strings

Use a hall reverb set to between 2.2 and 2.6 seconds with a predelay of at least 20ms (try doubling or tripling the time) timed to the track.

Using Modulation

Modulation is the third type of effect that adds dimension to a mix, although it accomplishes this more by movement in the soundfield than by ambience. Most musicians and engineers are very familiar with the types of modulation, but they're not clear on how they differ and when they're best used.

Types Of Modulation

There are three basic modulation effects used most often in a mix: phase shift, chorus, and flange. The differences between them are basically that a chorus and flange effect come as a result of a modulated delay that's mixed back into the original signal (with the flanger having shorter delay than a chorus) while, on the other hand, the phaser doesn't require a delay to achieve its effect (see Table 10.1).

All three modulation effects produce a series of frequency notches that slowly sweep across the frequency bands of a mix element, leaving only a series of peaks, which is what you hear. And here's where the differences are more pronounced, as a phaser has only a small number of notches spaced evenly across the frequency range while flangers and choruses have many more notches, spaced harmonically, which provides a much more aggressive sound.

Tremolo and vibrato are also popular modulation effects but they operate differently in that a delay isn't required for them to work. It's easy to get these confused, or even use their names interchangeably, but they are distinctly different effects: a tremolo varies the signal up and down in amplitude, while vibrato varies the pitch up and down (yes, even Fender got this wrong on its most famous amplifiers).

Table 10.1: The Differences between Modulation Effects

Effect	**Delay**	**Description**
Phase shift	None	Cancels out frequencies by shifting their phase to create the effect. Frequency notches are

		spaced evenly across the frequency range.
Flanging	0.1ms to 5ms	The deeper the frequency cancellations, the deeper the effect. Frequency notches are randomly and harmonically spaced across the frequency response.
Chorus	5ms to 25ms	Used to thicken the sound and create a stereo image. Frequency notches are spaced harmonically across the frequency response.
Tremolo	None	Cyclicly changes the volume.
Vibrato	None	Cyclicly changes the pitch.

Flangers And Phasers

The flanger is a dramatic effect that was first created in 1966 by Ken Townsend, the chief technician at EMI Studios in London (now known as Abbey Road Studios), in an attempt to come up with something called Automatic Double Tracking, or ADT. The effect was created because The Beatles' John Lennon loved the sound of his voice when it was doubled but hated the fact that he had to sing the song a second time, so the EMI tech staff was asked to come up with a solution. The ADT effect was accomplished by using two tape recorders at the same time and this led to an almost accidental discovery. Slowing one of the tape machines down ever so slightly by placing a finger on the tape flange (the metal part of the reel that holds the tape) created a sweeping harmonics effect. Lennon called this effect "flanging."

The first known use of flanging came on the Beatles' song "Tomorrow Never Knows," but one of the first big Top 40 hits that used the effect came a year later on a song called "Itchycoo Park" by the British group The Small Faces, which featured a large dose of the effect at the end of the song. Soon every artist and producer wanted flanging on their song, but there was a problem. In order to get the effect, two extra studio-quality tape recorders needed to be set up, which was both expensive and time consuming.

Even though technology was marching forward at breakneck speed, back in the 1970s the only feasible electronic simulation was an analog effect called a phaser, but the resulting sound had none of the same intensity of flanging. To this day, phasing isn't used very much because it's just not that dramatic an effect.

It wasn’t until digital delays came on the market in the 1980s that it became possible to replicate true tape flanging, and today just about every modulation plugin and stomp box can do at least a passable simulation of the effect if set up correctly.

Chorus

What makes a chorus different from a flanger is that the delay is longer, going from about 5 to 25ms, and the frequency notches needed to create the effect occur in a fixed cycle. The chorus effect shines in stereo and can really widen the sound of a track quite a bit. Since chorusing first was introduced by Roland in 1980, many hit-makers from that era used the effect over and over.

Today you’ll find that most modulation plugins and hardware allow you to easily switch between chorus, phasing, and flanging. They’re all related, but the ones that are used in stereo are the most dramatic.

Tremolo And Vibrato

For years, guitar amps included tremolo as standard feature, although in some cases (such as on Fender amps) it was mislabeled as vibrato. As stated previously, they’re not the same, since tremolo is a cyclic variation in volume, while vibrato changes the pitch of the sound.

Guitars weren't the only instrument to use tremolo, as both the original Rhodes and Wurlitzer electronic pianos had the effect built in. Vibrato is rarely used because the variation in pitch can make the track or other nearby tracks seem out of tune.

Typical Modulation Setups

While flanging, tremolo, and vibrato are pretty dramatic effects and can cause a mix element to stand out, chorusing is often used to widen a track. That said, modulation effects aren't usually set up in advance on a dedicated set of sends and returns, since the effect tends to work better on an individual instrument or vocal.

Nevertheless, a flanger is sometimes required across the entire mix, so it's best inserted either across the stereo buss or across a separate subgroup so the level can be easily automated and adjusted (see Figure 10.10).

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Figure 10.10: A flanger inserted on a subgroup

Modulation Techniques

Here are a few techniques often used when adding modulation to a mix element. Don’t be afraid to experiment though because any of these techniques may work well across a variety of mix situations.

For fatter lead or background vocals:

Use some chorusing panned hard left and right to fatten up the sound. Ride the chorusing effect, adding and subtracting it according to what sounds best.

For out-of-tune vocals:

If you have something against Auto-Tune or just want to cover up an out-of-tune vocal, use a stereo chorus or flanger and pan it slightly left and right. The more out of tune the vocal, the more modulation you need to cover it up. This does an effective job of taking the listener’s attention off of any sour notes.

Pan a delayed vocal behind the dry vocal and then send it to a chorus, detuning both sides a bit so the delay sounds wide and the modulation steals your attention from the tuning.

EQing Effects

One thing many mixers struggle with is getting reverb and delay effects to blend well in the mix. This happens more with reverbs than delays, especially during those times when the reverb just never seems to land quite right. Usually, the way the problem is addressed is to audition different reverb presets until something is found that just seems to work better, but that can take time, and it's easy to end up chasing your tail to the point where you're never sure which preset actually sounds the best. What many seem to forget is that most of those presets are the same basic reverb effect with different EQ settings, which you can add yourself to get there faster.

One thing that happened regularly back in the days of analog reverb (especially with plates) was to insert an EQ on the send before the actual reverb itself. Usually, the EQ was set more to cut than to boost — although you'd boost it if you wanted a bright-sounding plate that jumped out of the mix. Either way, if done well, the reverb would suddenly fit a lot better in the track. In fact, back in the classic days of the big studios, this was done in the back room and not left up to the engineer at the console. In fact, it became one of the reasons for clients wanting to work there; they loved the sound of their reverbs.

We can use those same techniques today using reverb plugins on our DAW, or any other effects for that matter. Just remember that it usually

sounds best if the EQ is placed before the reverb, not after it, because it activates the plugin differently so it has a greater effect on the frequency response of the reverb.

TIP: Try adding tape-saturation plugins such as Avid's Heat or Universal Audio's Studer A800 Tape Recorder to the effect send or return. The extra harmonics sometimes give it more depth.

The following sections describe three EQ curves that are frequently used.

On Vocals

One thing about reverb is that any low end that it has just muddies up the track, and any high end may stick out too much, which is why it's often a good idea to roll off each end of the frequency spectrum a bit. In many cases, this means somewhere around 200Hz and 10kHz (or even lower). When reverb is used on vocals, sometimes it fits better if there's also a bit of an EQ scoop in the midrange around 2kHz, where the consonants of the vocal live, so the effect stays out of the way frequency-wise (see Figure 10.11). Once again, this is very effective on delays and modulators as well.

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Figure 10.11: Effects EQ curve for vocals - HPF at 600Hz, dip at 2kHz, LPF at 10kHz

On Instruments

For instruments, the Abbey Road curve, which is what the famous studio has used on their reverbs since the 1960s, works very well. In this case, the low end is rolled off at 600Hz and the high end at 10kHz (see Figure 10.12). This curve makes any reverb sound a lot smoother and fit better with the track. You'll find that this setting just increases the depth without it sounding washed out when you add more reverb using this curve. Of course, too much of a good thing is no good either, so be judicious with the amount you add.

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Figure 10.12: The “Abbey Road” effects EQ curve - HPF at 600Hz, LPF at 10kHz

On Drums

Sometimes, calibrating the reverb on the drums is the toughest of all because you want depth without calling attention to the ambience. A good way to do that is a variation of the Abbey Road curve where the high end is severely rolled off to 6k, 4k, or even 2kHz (see Figure 10.13). You’ll find that you’ll have some depth without the ambience ever calling attention to itself.

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Figure 10.13: Effects EQ curve for drums - HPF at 600Hz, LFP at 6kHz

While these EQ curves work great with reverbs, don't be afraid to try them with delays or modulation effects as well, because the results are very similar. You'll get depth without the delay or the effect getting in the way. Of course, if you want to really hear the reverb or delay, go the opposite way and increase the high end, and the effect will jump right out of the track.

Equalization Tips For Reverbs And Delays

To make an effect stick out, brighten it up.

To make an effect blend in, darken it up (filter out the highs).

If the part is busy (such as with drums), roll off the low end to keep it out of the way.

If the part is open, add low end to the effect to fill in the space.

If the source part is mono and panned hard to one side, make one side of the stereo effect brighter and the other darker and pan the brighter

side opposite the source track.

Layering Effects

Layering means that each mix element sits in its own ambient environment and each environment is artificially created (or enhanced) by effects. The idea here is that these sonic atmospheres don't clash with one another, as in the case with instrument frequency ranges.

The following section features some suggestions so that the sonic environments don't clash.

Layering Tips for Reverbs and Delays

Layer reverbs by frequency, with the longest being the brightest and the shortest being the darkest, or the opposite. Pan the reverbs any way other than hard left or right.

Return the reverb in mono and pan accordingly. All reverbs needn't be returned in stereo.

Make elements big with reverbs and add depth with delays, or vice versa.

Use a bit of the longest reverb on all major elements of the track to tie all the environments together.

"My personal taste is to use more layers, like using several reverbs to create one reverb sound, or using several short and long delays. My reverbs and effects usually end up coming from four to eight different sources. They'll be short, long, bright, dull, and everything you need to make an environment."

—Bob Bullock

Re-amping

One of the ways to re-create a natural environment is by a process known as re-amping. This is accomplished by actually sending the signal of an already-recorded track (say a guitar or keyboard) back out to an amplifier in the studio, then miking it from a distance to capture the ambience of the room (see Figure 10.14). This works best if the ambience is captured in stereo.

Figure 10.14: Reamping to achieve natural ambience

"What I will do frequently when we're layering with synths is to add some acoustics to the synth sounds. I think this helps in the layering in that the

virtual direct sound of most synthesizers is not too interesting, so I'll send the sound out to the studio and use a coincident pair of mics [X/Y stereo pair] to blend a little bit of acoustics back with the direct sound. It adds early reflections to the sound, which many reverb devices can't do."

—Bruce Swedien

11

The Interest Element: The Key To Great (As Opposed To Merely Good) Mixes

Applying the previous five elements may be sufficient for many types of projects but a hit song requires that a mix be taken to another level. Although it's always easier when you start your project with great tracks, a solid arrangement, and spectacular playing, a great mix can take even run-of-the-mill tracks and transform them into something so compelling people can't get enough of it. This isn't just a theory; it's been demonstrated time and again on some of your all-time favorite songs.

So how can we get a mix to that point?

More than being just technically correct, a mix must be as interesting as a good movie. It must build to a climax while having points of tension and release to keep the listener subconsciously involved. Just as a film looks bigger than life, a great mix must sound bigger than real life. The passion and the emotion must be on a level where the listener is sucked in and forced to listen.

Which brings us back to where we started:

"The tough part, and the last stage of the mix, is the several hours it takes for me to make it sound emotional and urgent and exciting so that it's not just a song, it's a record. It's not making it just sound good, it's making it sound like an event. That's the last stage of when I mix, and that's the part that makes it different or special."

—Ed Seay

The Direction Of The Song

The first thing that the mixer does before diving headfirst into the mix is find the direction of the song, and that's determined by both the style of the artist and the performances by the players. For instance, if the song is folksy in nature, then it probably won't need big, bombastic drums and long reverbs and delays, but if the artist is a loud arena rock band, then you probably won't want a close, intimate sound.

It's absolutely possible to change the direction of the song and have a hit, although it's usually based on the style of the artist. A good example of this is Marvin Gaye's classic "Heard It through the Grapevine," which has been a hit by many artists in innumerable styles through the years. The direction of Creedence Clearwater is very different from the direction of Gladys Knight and the Pips, the Kaiser Chiefs, or Amy Winehouse, yet it works equally well for all of them. The direction is a function of the artist and the performance.

Develop The Groove

All good music, regardless of whether it's rock, jazz, classical, rap, marching band, or some new space music that we haven't heard yet, has a strong groove. So what exactly is the groove?

The Groove Is The Pulse Of The Song

A common misconception about a groove is that it must have perfect time. A groove is created by tension against even time. That means that it doesn't have to be perfect, just even, and that all of the performances don't have to have the same amount of "even-ness." In fact, it makes the groove stiff-feeling if the performances are too perfect. This is why perfect quantization of parts and lining every hit up to a grid in a DAW frequently takes the life out of a song. It's too perfect because there's no tension. It's lost its groove.

Just about every hit record has a great groove and that's why it's a hit, but if you really want to study what a groove is, pick any song from one of the masters: James Brown, Sly Stone, George Clinton, or Prince. Every song contains the audible essence and feel of a groove.

We usually think of the groove as coming from the rhythm section, especially the drums, but that's not necessarily always the case. In the Police's "Every Breath You Take," the rhythm guitar establishes the groove, while in many songs from Motown's golden age by the Supremes, the Temptations, Stevie Wonder, and the Four Tops, the

groove was established by James Jamerson's or Bob Babbitt's bass. It's even been said that Michael Jackson's vocal was a groove unto itself and a song could easily be built around it.

The trick for the mixer is to find what instrument best defines the song's groove and then build the rest of the mix around it.

Finding The Groove

Regardless of which instrument is providing the groove of the song, if you want a great mix, you've got to find it before you do anything else. Have a listen to a rough mix of the song or just push all the faders up and see whether you can feel its pulse.

Ask yourself the following:

What instrument (or instruments) is providing the pulse? Is it the drums? The bass? A keyboard? A loop?

What would make the groove stand out? The balance? The tone? Compression? An effect?

Is there a rhythm arrangement element to the mix, such as percussion, a rhythm guitar, a keyboard, or a loop (check back to Chapter 5)? Is it essential to the groove?

Building The Groove

While it's true that the groove may be the result of a single instrumental performance, usually it's built around the interplay of different instruments, especially in complex mixes with a lot of tracks.

Normally, the groove of the song is provided by the bass and drums, but it's important to determine whether another instrument, such as a rhythm guitar, keyboard, loop, or percussion, is an integral part that makes up the pulse of the song. Usually this can be easily identified as an instrument that's playing either the same rhythmic figure as the bass and drums or a multiple of the rhythm, such as double time or half time.

After those additional rhythmic elements are discovered, here's one way to build the groove:

Find the mix element that provides the basic pulse of the song (frequently the drums).

Add the lowest-frequency instrument that's playing the same or a similar rhythmic figure (usually the bass).

Add any additional instruments playing the same or a similar rhythmic figure in order of frequency, from low to high.

Add any instrument playing a similar rhythmic figure, such as half or double time.

Add any instrument working as the rhythm arrangement element (remember the section about arrangement elements in Chapter 5?) and providing motion to the song (such as a shaker or tambourine).

The groove may be attributed to only a single instrument, such as in the case of a power trio (guitar, bass, and drums), to three or even four instruments on rare occasions. If you’re not sure, the best way to determine what’s playing the groove is to try mixing in different combinations of instruments along with the rhythm section to see whether the pulse gets stronger, gets weaker, or stays the same.

TIP: If a new instrument adds to the pulse of the song and the pulse seems lessened if it’s muted, you’ve found an instrument that’s an essential part of the groove.

Find The Most Important Element And Emphasize It

Equally as meaningful, and in some cases even more important than the groove, is finding whatever element is the most important to the song. In some cases (such as EDM, dance, and hip-hop music), the most

important element is the groove, yet in other genres it may be the vocal (such as in country or pop).

Even though the most important element is often the lead vocal, it doesn't necessarily have to be. It could be a riff, such as the intros to the Stones' "Satisfaction" and "Start Me Up," the piano line from Coldplay's "Clocks," or the guitar line from the White Stripes' "Seven Nation Army." It's always a part so compelling that it forces you to listen and re-listen to the song.

Whatever element is determined to be most important, the mixer must identify and emphasize it in the mix in order for the song to be elevated beyond the ordinary. Like most other creative work, that requires inspiration, but you can't underestimate the value of talent and experience in the process.

To emphasize the most interesting element, try the following:

After the element is determined, raise its level a dB and then increase it by 1dB steps. Does the part jump out of the mix?

If the part isn't compressed, insert a compressor and begin by adding 2dB of compression. If the part is already compressed, add an additional dB and increase in 1dB steps. Be sure to compensate for the decrease in level by adjusting the Output control. Does the part jump out of the mix?

If the part is already EQ'd, add an additional dB at the EQ points previously selected. If the part isn't EQ'd, insert an equalizer and add 1dB at 5kHz. Does the part jump out of the mix now? What if you add a dB or two at 1kHz?

If the part is dry, try adding some effects, starting with any reverb or delay that's already set up and in use in the mix. If that doesn't work, add a dedicated reverb beginning with the parameters set from short to long. Move on to delay, modulation, distortion and any other effects at hand. Does the part jump out of the mix now?

"I try to find what's important in the mix. I try to find out if the lead vocal is incredibly passionate and then make sure that the spotlight shines on that. If, for instance, the mix needs 8th notes, but they're going [makes sound effect] and it's not really pushing the mix, sometimes playing with compression on the acoustics [guitars] or auditioning different kinds of compression to make it sound like, 'Boy, this guy was into it.' It's just basically playing with it and trying to put into it that undefinable thing that makes it exciting. Sometimes hearing the guy breathe like the old Steve Miller records did. With a little of that, you might say, 'Man, he's working. I believe it.' It's a little subconscious thing, but sometimes that can help."

—Ed Seay

Fifteen Steps To A Better Mix

Mixing is a nebulous art in that most musicians and engineers learn more by feel and listening than by being taught. As a result, a number of important items in a mix can be easily overlooked, and these can

mean the difference between a mix that sounds polished and professional and one that sounds amateurish.

Here's a checklist of items that can help you think in a little more detail about your mix and tighten it up as a result.

Does your mix have dynamic contrast? Does it build as the song goes along? Are different instruments, sounds, or lines added or muted in different sections?

Does your mix have a focal point? Is the mix centered around the instrument or vocal that's most important?

Does your mix sound noisy? Have you eliminated any count-offs, guitar amp noises, bad edits, and human breaths that stand out? Each one may not seem like much, but their effect is cumulative.

Does your mix lack clarity or punch? Can your ear distinguish every mix element? Does the rhythm section sound great by itself? Is the balance between bass, kick, and snare correct?

Does your mix sound distant? Try decreasing the level of the reverb and effects, starting with the wettest and then working your way to the driest.

Can your hear every lyric? Every word must be heard clearly. Tweak the automation if you’re using it; automate the track if you’re not.

Can you hear every note being played? If solos or signature lines are being masked, automate the track to hear every note, or tweak the automation until you can.

Are the sounds dull or uninteresting? Are generic synth patches or predictable guitar or keyboard sounds being used? Try modifying them with an effect.

Does the song groove? Does it feel as good as your favorite song? Are the instrument or instruments that supply the groove loud enough?

What’s the direction of the song? Should it be close and intimate or big and loud? If your current direction isn’t working, try the opposite.

Are you compressing too much? Does the mix feel squashed? Is it fatiguing to listen to? Is all the life gone? If so, decrease the mix buss compression first. Then decrease the main instrument or vocal compression. Decrease the rhythm-section compression next and, if these fixes don’t help, decrease the compression on everything else.

Are you EQing too much? Is it too bright or too big? Decrease the upper midrange EQ on the vocals, guitars, loops, and snare. Decrease the low-frequency EQ on the bass and kick.

Are your fades too tight? Does the beginning or ending of the song sound clipped? Adjust the fades until the attack transient of the downbeat is distinct.

Did you do alternate mixes? Did you do at least an instrumental-only mix?

Did you document the keeper mixes? Are all files properly named? Are you sure which file is the master? Have you made a backup?

An interesting mix is all in the details, and details take time to sort out. Working through each one of these steps may take a while, but the end result will be worth the effort.

Advanced Techniques

Whereas a decade ago a mix was deemed complete as soon as it felt good to everyone involved, mixes these days require more precision than ever before. This is because the mindset of a mixer has changed thanks to the ability to now perform many more mix moves in a repeatable manner inside a DAW. In this chapter we'll look at a few of the more advanced techniques used to make a mix competitive today.

Keep in mind that cleanup, timing adjustment, and pitch correction are considered more a part of production and should be completed before the track is sent to be mixed, but mixers are sometimes expected to perform these tasks anyway.

Cleanup

As stated in Chapter 3, a big part of a mixer's responsibility can be track cleanup. Although it's always preferable for this to take place prior to mixing, sometimes the previous engineers don't have time, are careless, or maybe even don't know how to properly clean up the tracks before mixing begins.

What do we mean by cleanup? It's simple. The tracks need to have the least amount of aural distraction before the mix can start. It's important that any clicks or pops are removed, bad fades are fixed, and any random, accidental or otherwise unnecessary noises are eliminated. Here's how to do it.

Removing Noise

Noise on a track can mean anything from the guitar-amp buzz before the guitarist plays a part, to the shuffling of feet or the clearing of the throat captured by the vocal mic before the singer begins to sing, to even breaths in between vocal phrases. As a matter of course, the length of each audio clip should be trimmed to just before and after the part plays to eliminate the noise (see Figure 12.1). Be sure to add a short fade-in or fade-out to eliminate any potential clicks (see Figure 12.2).

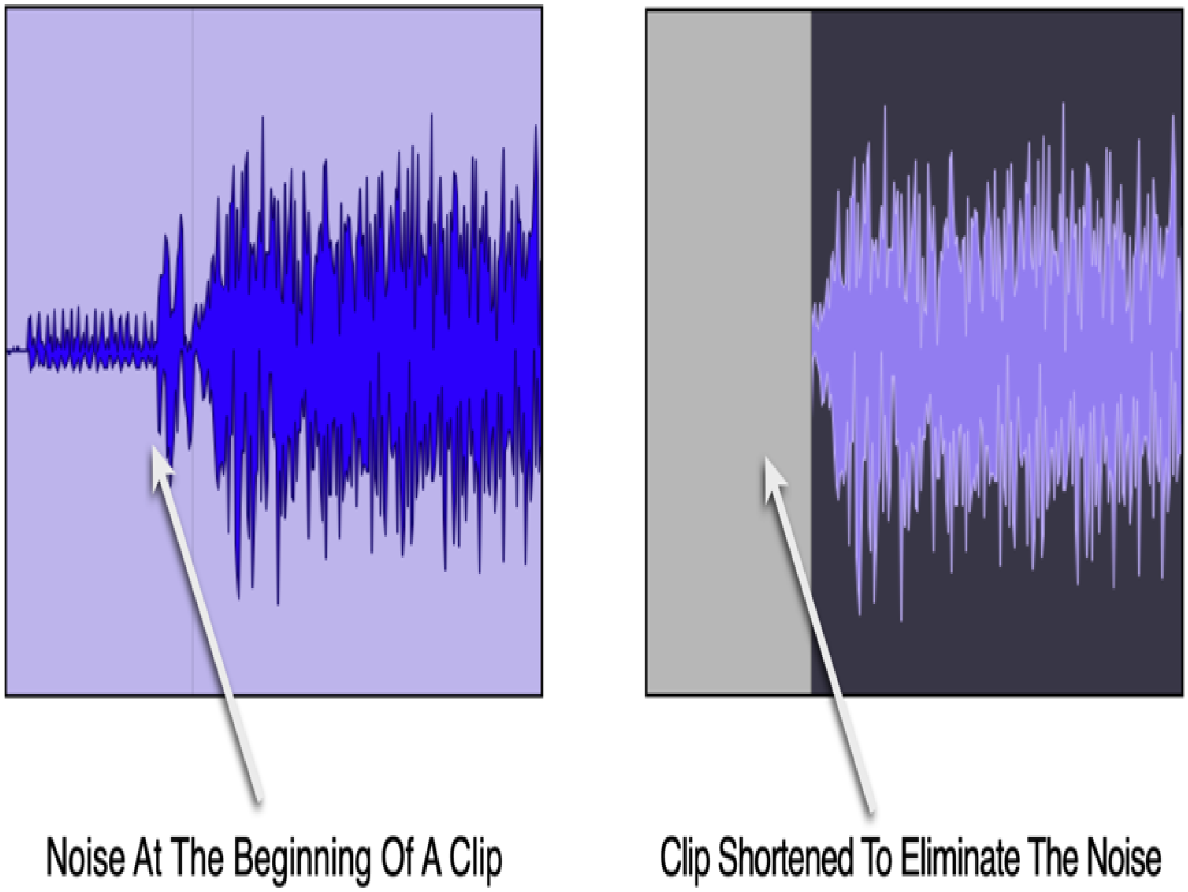


Figure 12.1: Shortening the clip length to eliminate noise

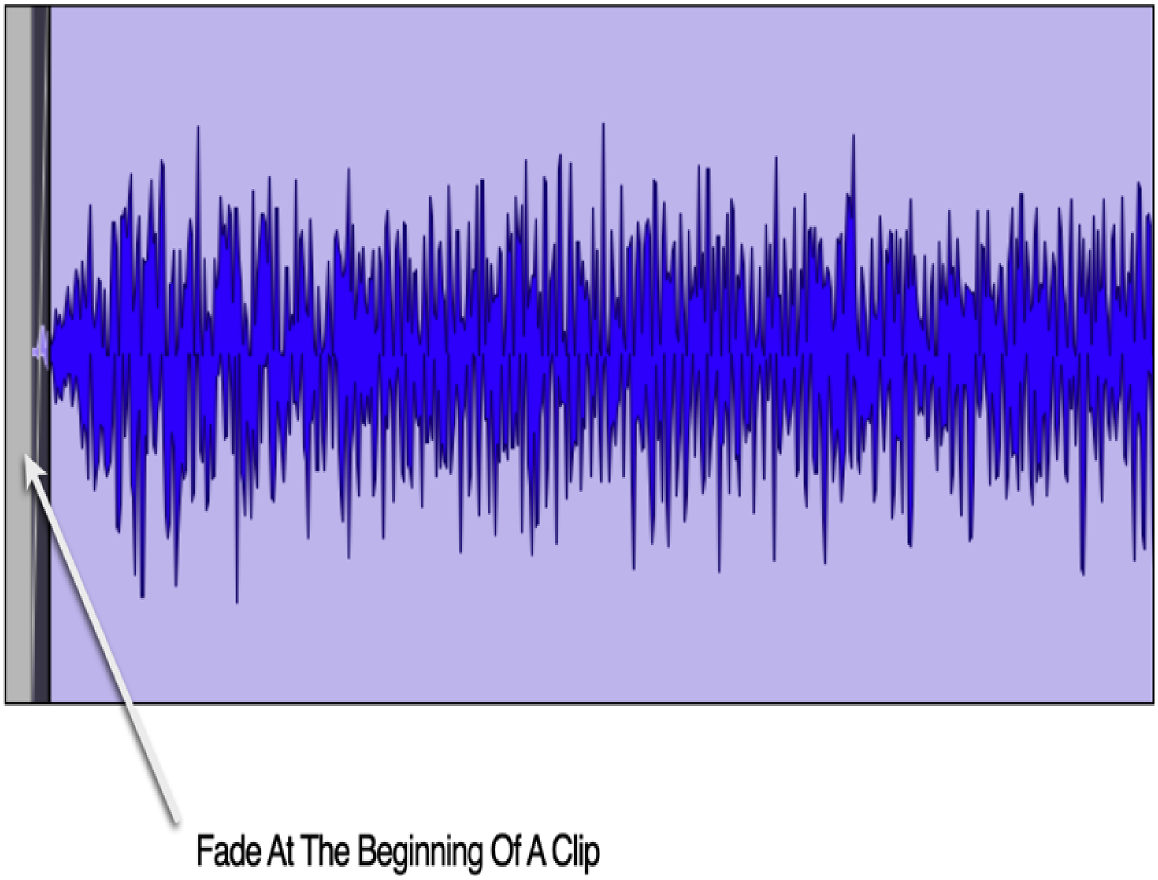


Figure 12.2: Adding a short fade to eliminate clicks

While shortening the clip doesn't usually pertain to the drum track because doing so can make it sound choppy and unnatural, there are times when it's appropriate, such as in the case where there's a measure or more of silence in the song. Be sure to use a fade that both sounds natural and doesn't cut off the cymbal decay or the attack of the next downbeat after the silence (see Figure 12.3).

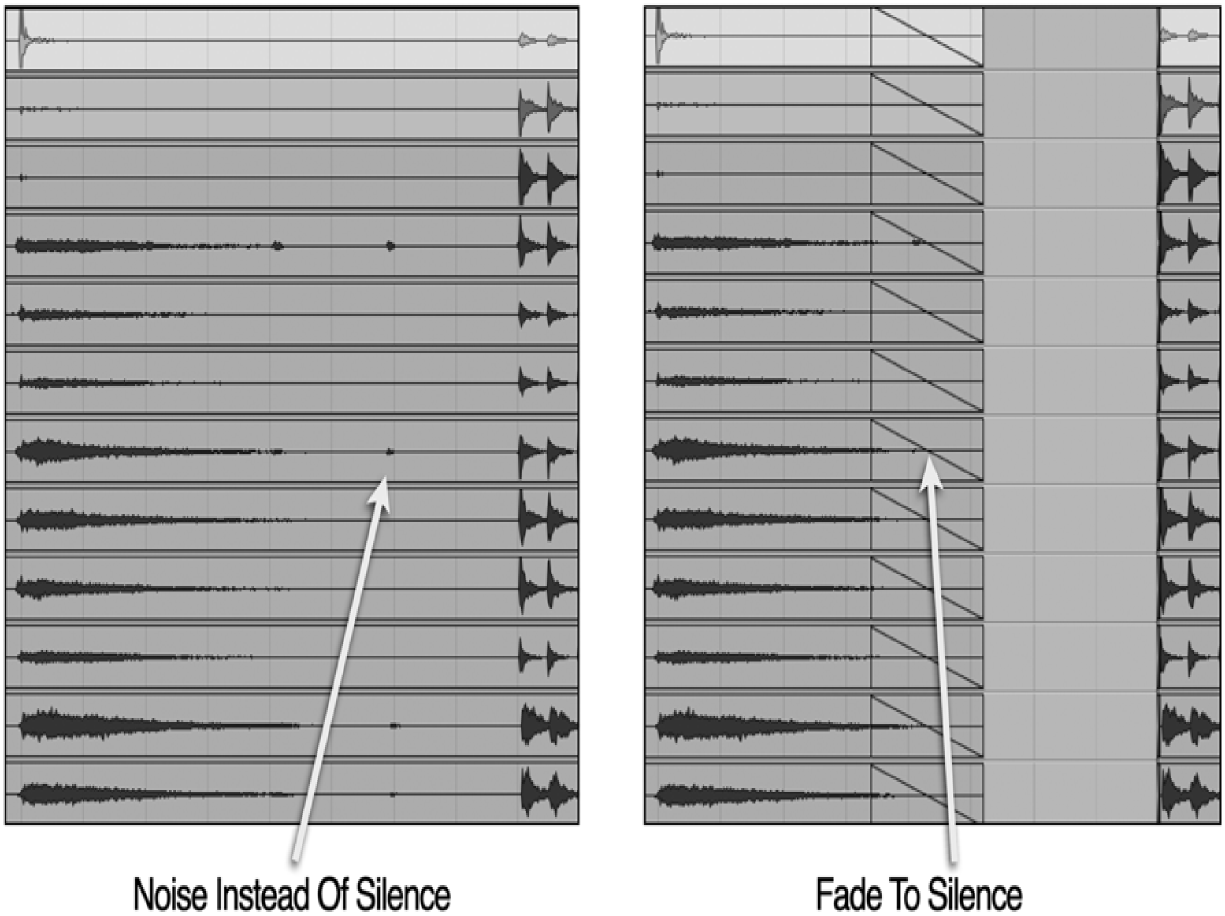


Figure 12.3: Adding fades to a drum track

Removing Clicks And Pops

Clicks and pops can come from a number of situations, including butt-cut edits (edits without fades that are butted up against one another) or transient overloads, such as from a thumb-slapped bass with active

pickups. Most clicks occur during cut and paste, timing adjustment, or noise elimination operations where a fade isn't used so only a butt-cut remains (see Figure 12.4). Although the click is sometimes very apparent, many times it's quiet enough that it will get lost in the track when the other instruments are introduced. Sometimes an edit can be completely silent if it happens to land in the right place of the audio waveform.

Figure 12.4: A butt-cut edit

TIP: While you may not hear the clicks when the track is played back over monitors, you may find they jump right out when you're listening with headphones. This is why it's best to always listen to a complete pass with phones to see whether any clicks are audible.

The way to fix any potential trouble spots is by adding fades to the edits. This is something that you can do by eye, although it's time consuming. While you may hear clicks on butt-cuts at the beginning and ending of clips, it's more likely that the ones that will stick out are edits that lack crossfades. Go through the track and add crossfades where appropriate (see Figure 12.5).

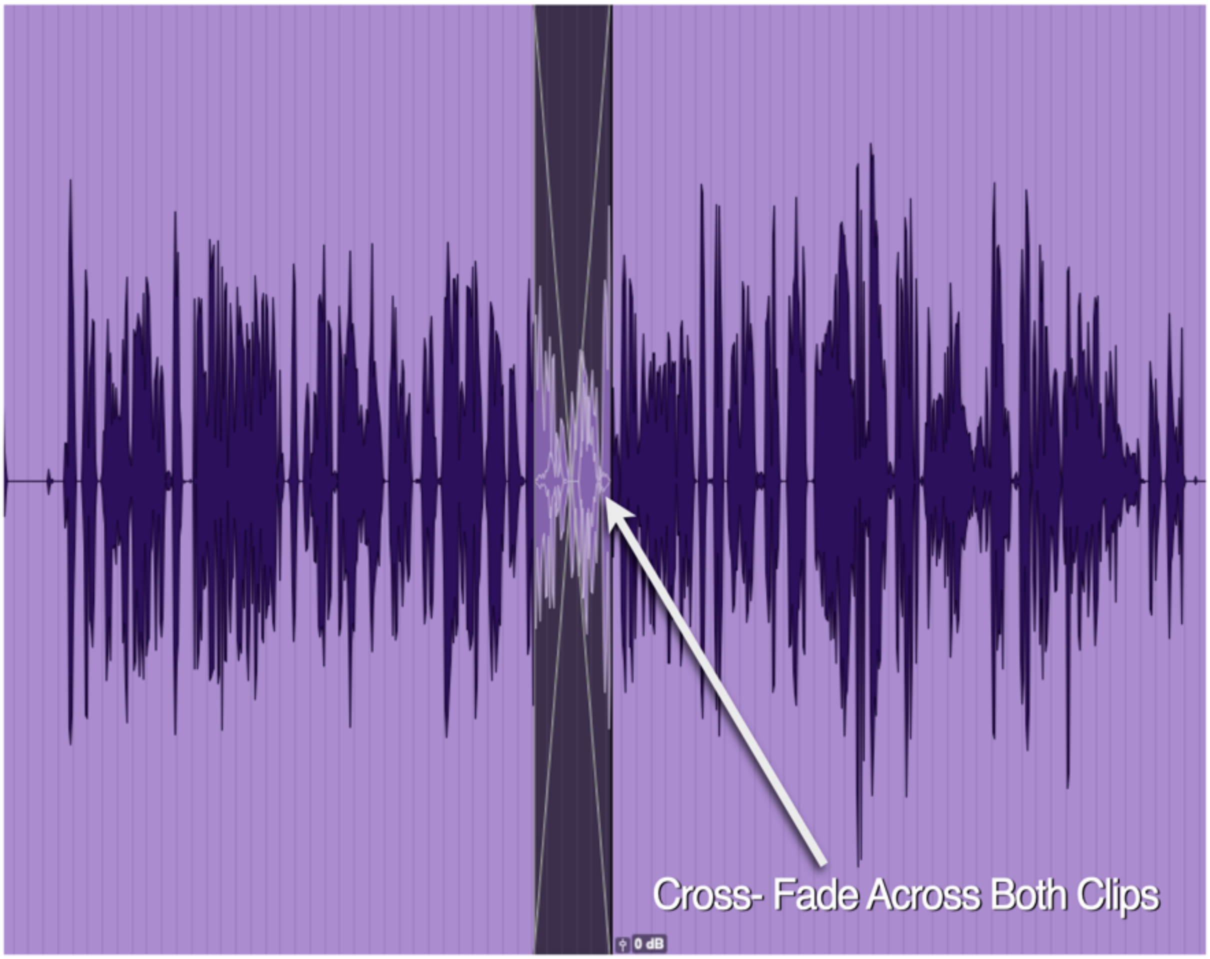


Figure 12.5: Adding a crossfade

Remember that even though these clicks might not be readily apparent, any kind of noise accumulates and plays a very subtle role in how the listeners perceive the song. Sometimes they can't explain the difference, but they can hear it.

Removing Count-Offs

Leaving in count-offs such as drumstick clicks (some musicians call them “count-ins”) is a sure sign of a demo recording and it’s something no one wants to listen to. While you can set any DAW to begin precisely on the downbeat of the song after the count-off, it’s best to eliminate the count-off altogether, especially if there’s another instrument that’s beginning the song over what’s supposed to be silence.

There are two ways to eliminate a count-off: either shorten the clip length as we did previously for noise elimination, or make a cut so there’s a separate clip containing the count and then mute it (see Figure 12.6). Muting the clip leaves the count intact which might be useful in the event it’s needed later for additional overdubs (although this shouldn’t be the case once it’s time to mix).

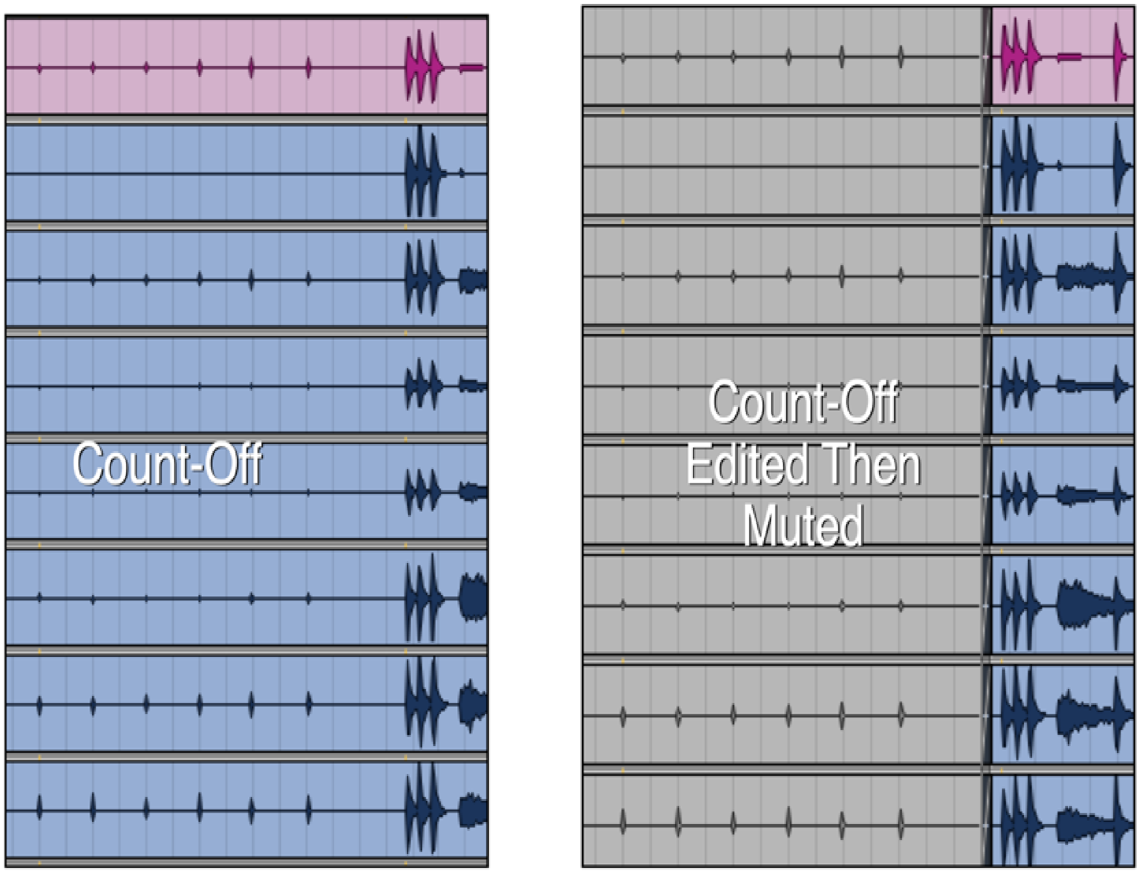


Figure 12.6: Creating a separate count clip and muting it

Fixing Bad Fades

Sometimes adding a default fade just doesn't sound natural. Maybe the fade is too tight and cuts off the attack or release of the part, or the fade just doesn't sound smooth enough. Now is the time to fix any fades that don't work in the track by adjusting the fade timings to solve the problem as intended. (see Figure 12.7).

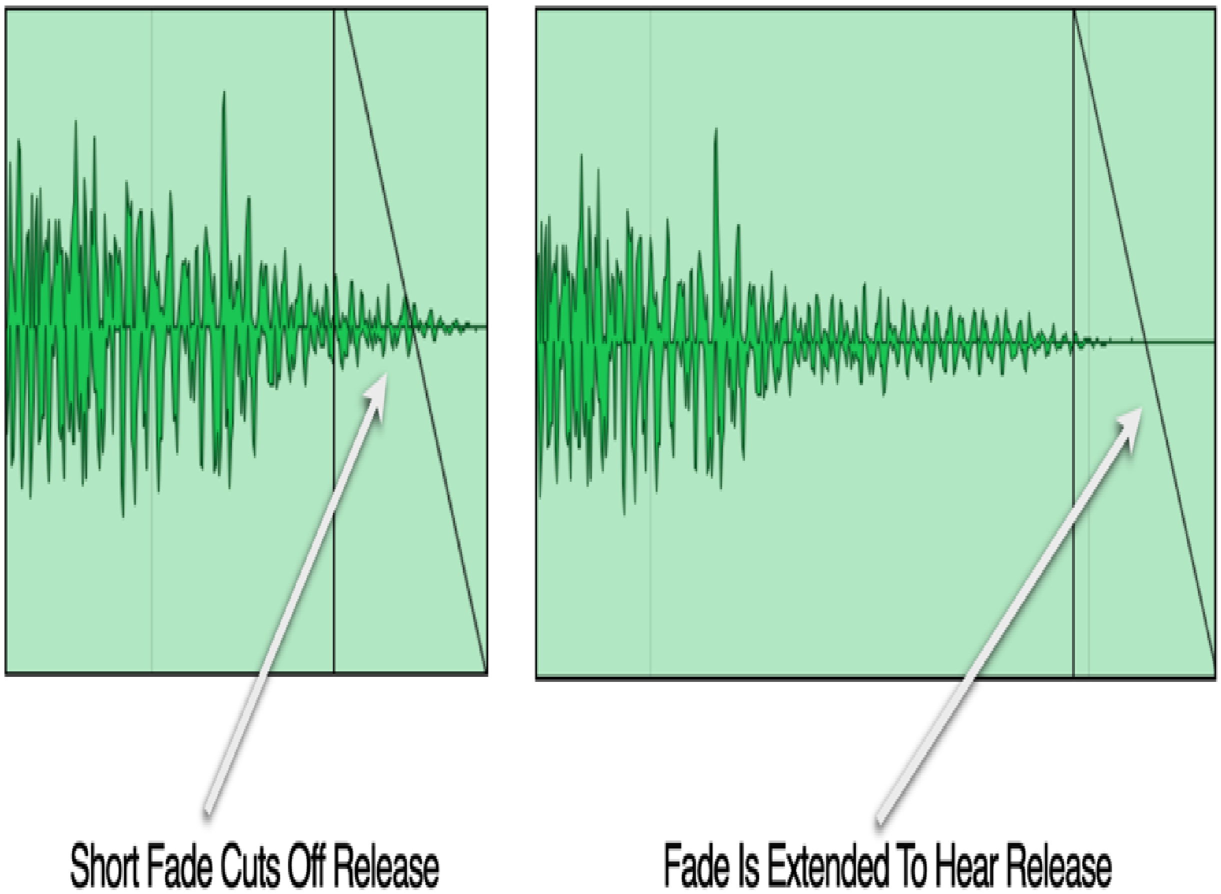


Figure 12.7: Fixing a short fade

If you have a fade that seems unnaturally fast at the end (especially a fade-out), try an exponential power fade instead of the default linear fade (see Figure 12.8).

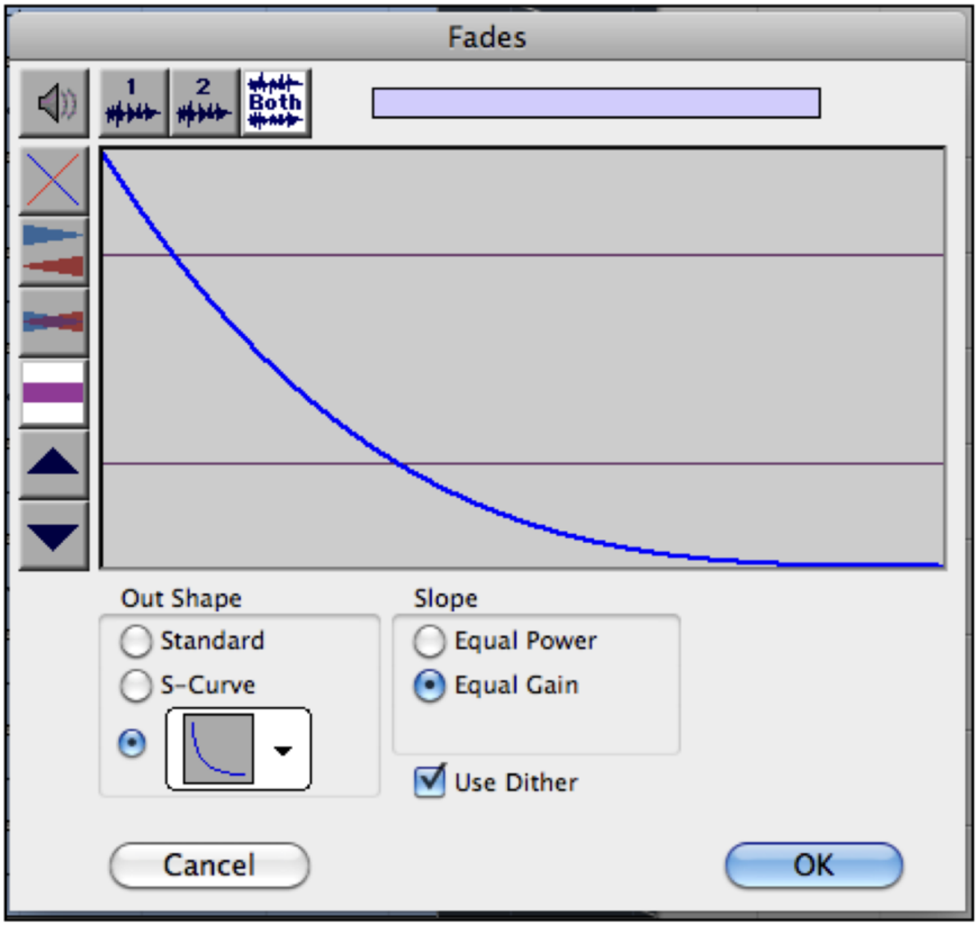


Figure 12.8: An exponential power fade

Eliminating Unwanted Distortion

Distortion can be anything from a breath blast to a note on a direct bass that's been so hot it causes an overload and clips the waveform. In both cases it's the transient of the attack that causes the problem but

thankfully it's extremely short. There are a number of ways to either fix it or mask it.

Replacement

Sometimes the most natural sounding operation is to replace the area of distortion with a similar area from another part of the song that's clean. That means finding a similar piece, copying it, and then pasting it over the section of the clip that has the distortion or noise (see Figure 12.9).

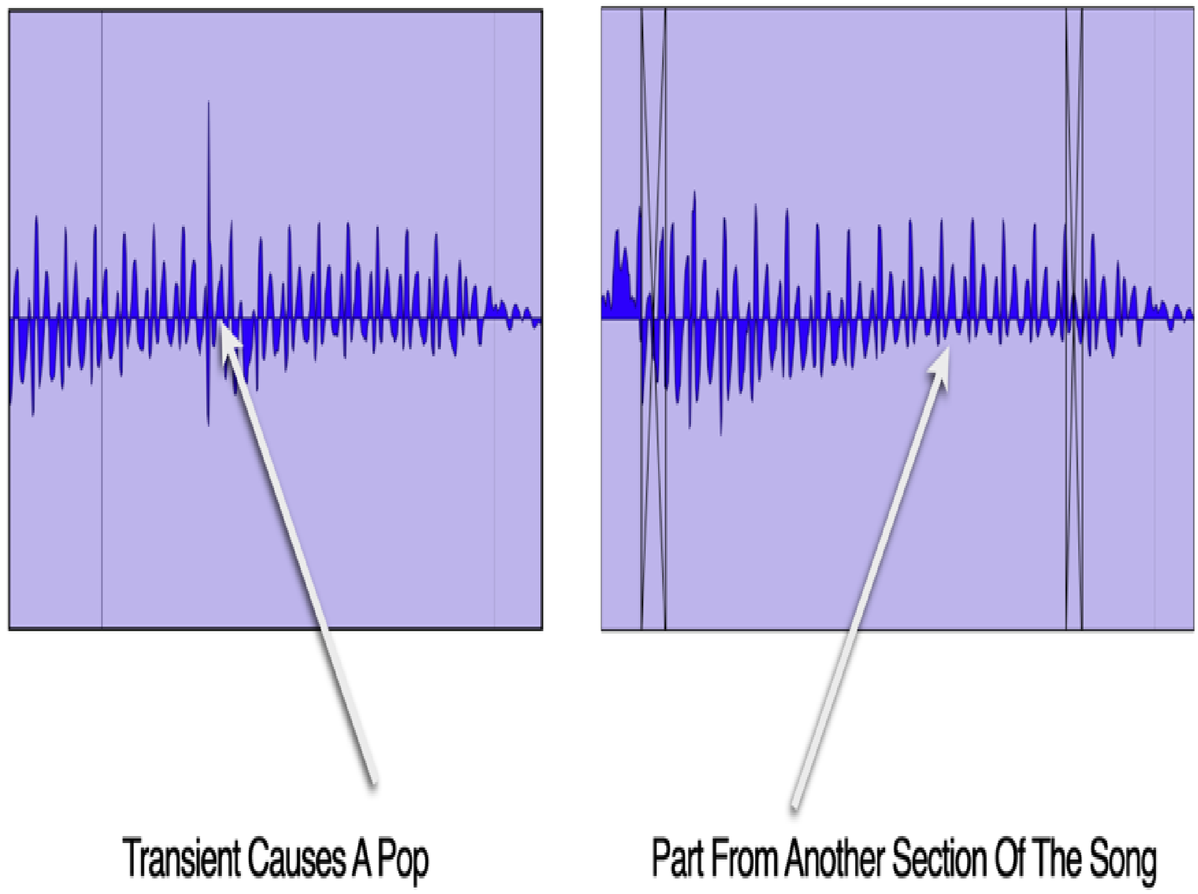


Figure 12.9: Copy and pasting to eliminate distortion

Clip Level Adjustment

If a clean section of the song isn't available to copy and paste, the next thing to try is to adjust the level of just the transient. Make a cut around the transient and lower the level using Clip Gain until it sounds natural (see Figure 12.10). This feature may not be available on all DAWs.

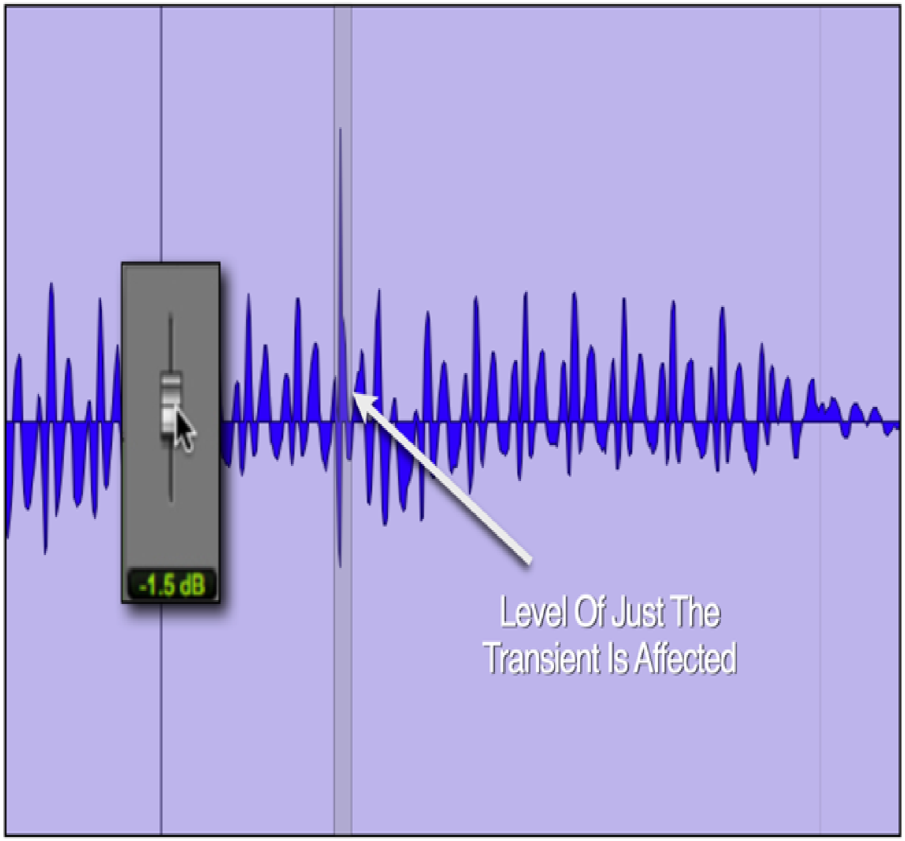


Figure 12.10: Treating the transient with a clip level adjustment

Automation

If a clip level adjustment feature isn't available on your DAW, you can accomplish the same thing using track automation. Draw the automation curve so the level of the transient decreases to a point where it seems natural (see Figure 12.11). You can read more about automation techniques later in this chapter.

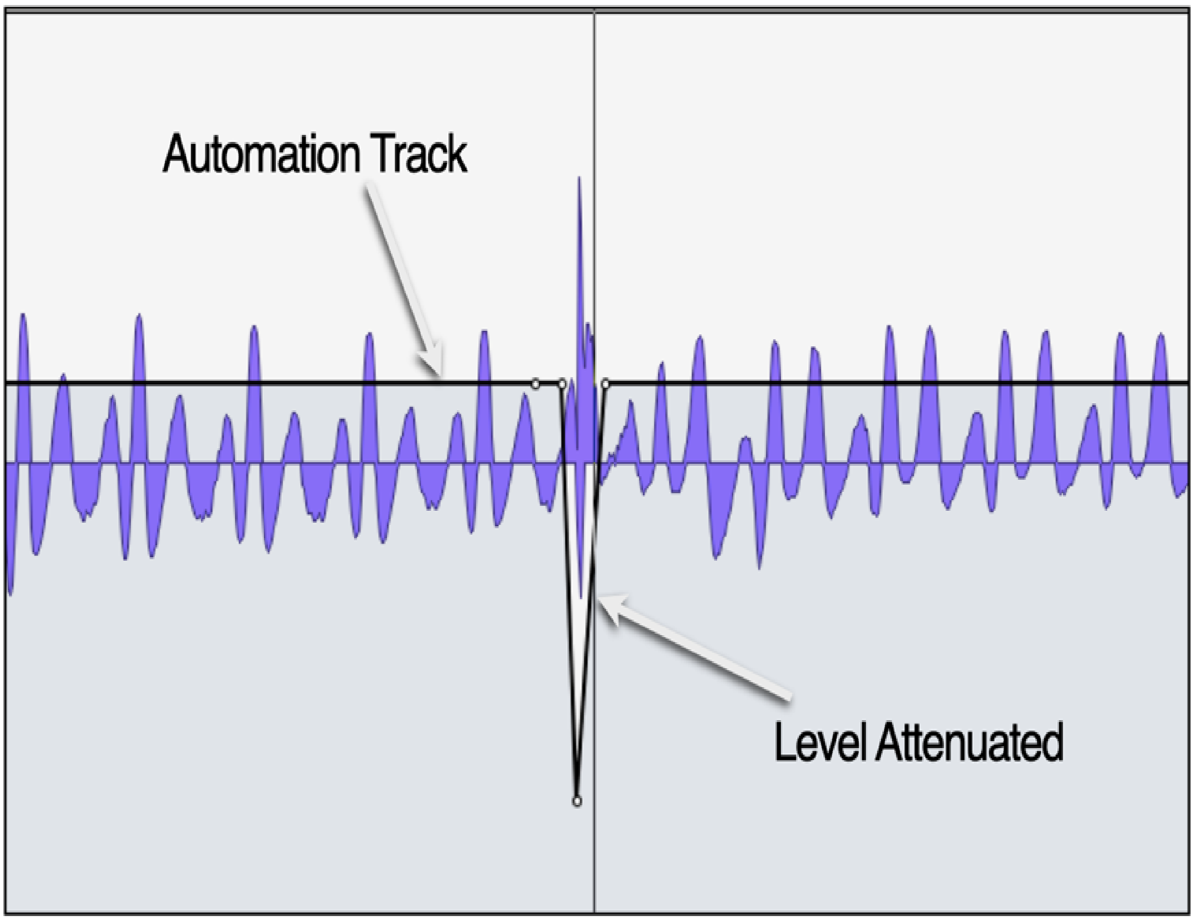


Figure 12.11: Using the automation to eliminate distortion

Elimination

Sometimes a transient passes so quickly that you can easily delete it without it being noticed. Be sure to add fades to the new edits so as to not create a new click (see Figure 12.12).

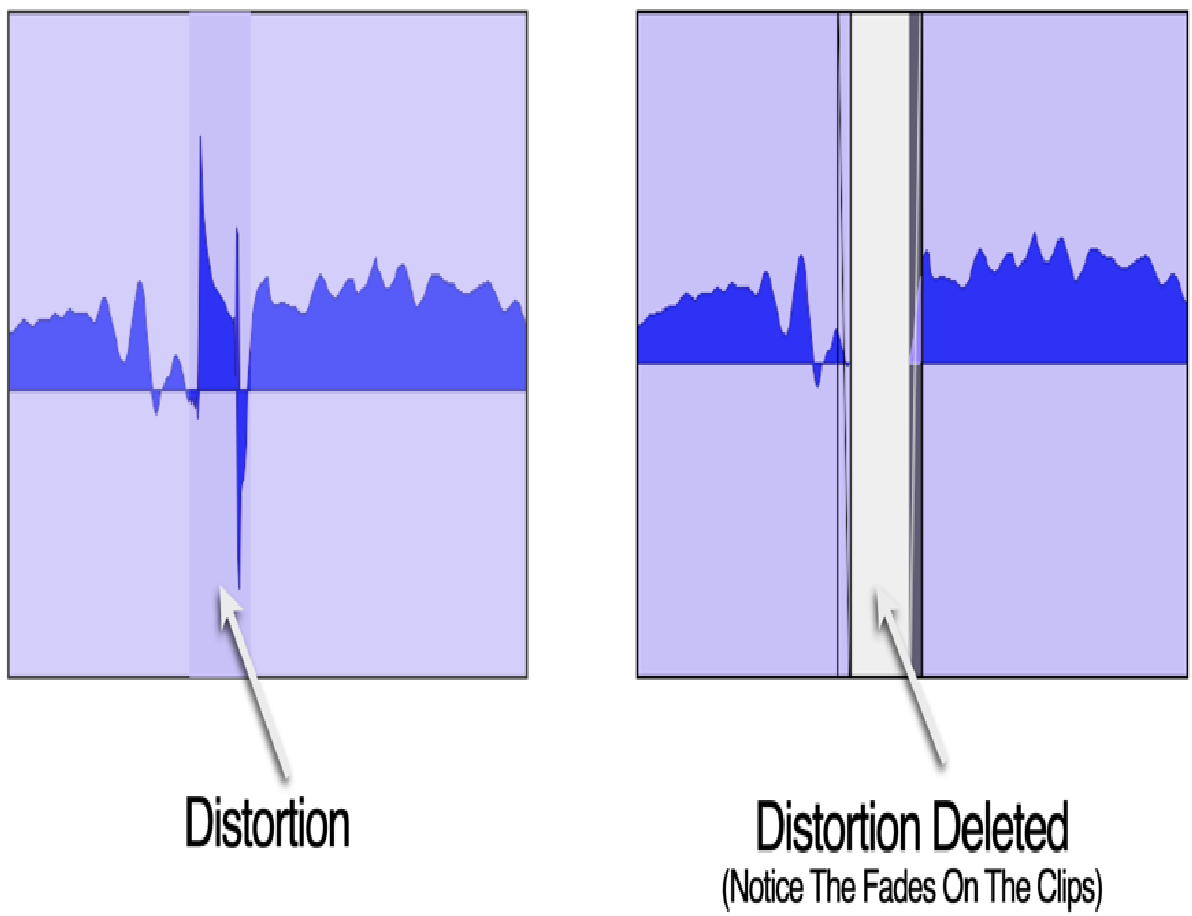


Figure 12.12: Deleting a transient

Deleting Extra MIDI Notes

Delete any extra “split” notes that were mistakenly played when the part was originally recorded. You might not hear these extra notes distinctly when all the instruments are playing together, but just like the noises at the beginning of individual tracks, they have a tendency to rise to the surface as things get compressed (see Figure 12.13), Just like with the other noises we discussed, these things are all cumulative.

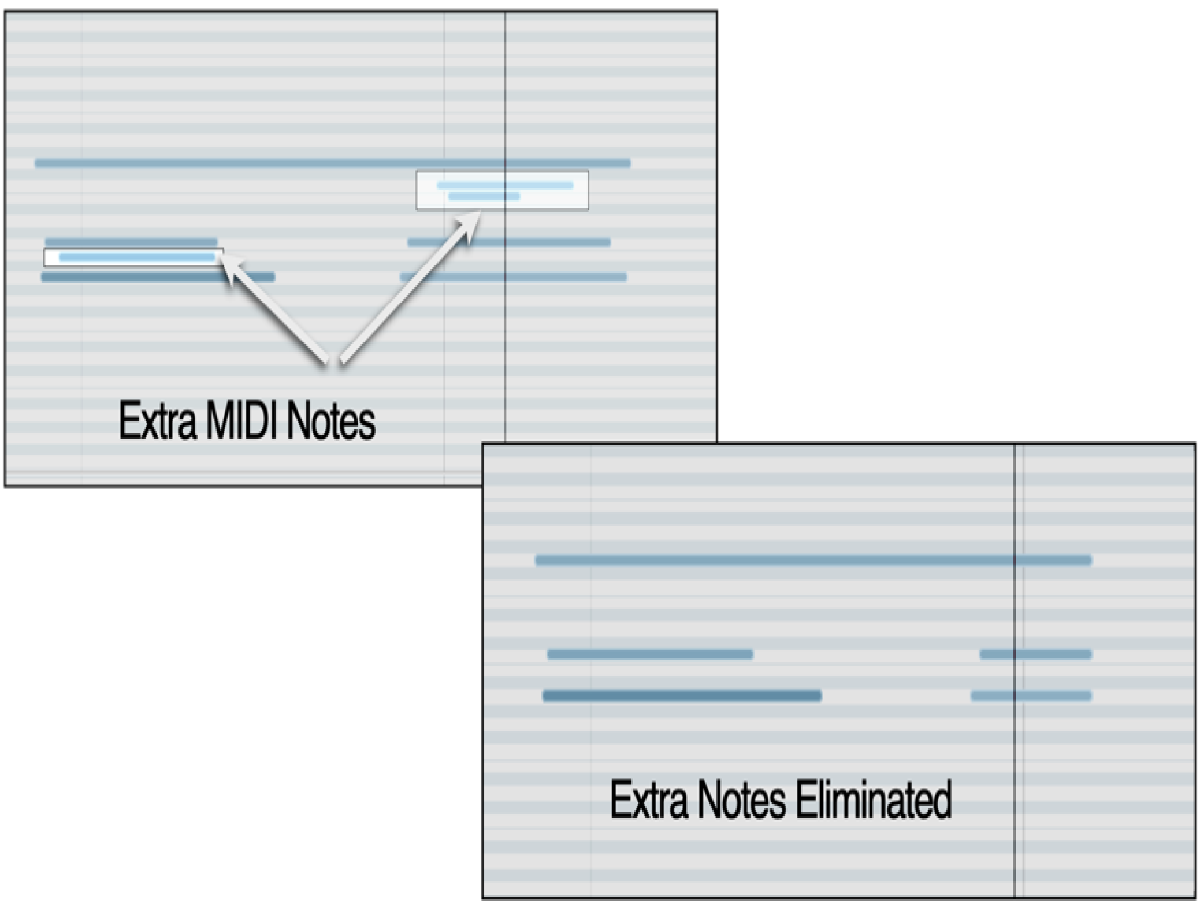


Figure 12.13: Deleting extra MIDI notes

Adjust The Timing

As we discussed at the top of this chapter, mix engineers are called upon to do much more than ever before. Individual track editing used to be done by the producer before the tracks were ever sent to for mixing, but mixers now find it's a job they're expected to do.

Even with great A-level players on a session, there's always some portion of a recording that doesn't feel quite right. Usually, the timing of the basic tracks will be tweaked right after your tracking session (if you're not using loops or beats or MIDI drum samples) so you have a solid rhythm section to overdub against, but if you're just now discovering some sections of an overdub that don't feel right (which happens more than you might think), prepare for the joys of slipping and sliding time.

Here's a list of dos and don'ts when tweaking the track timing:

Don't edit by eye. For most music (electronic music being the exception), you can't successfully time your tracks by lining everything up in your DAW using the kick and snare or even the grid as a visual reference and expect it to sound natural and organic. More often than not, tracks that look perfectly lined up don't sound or feel right, which

is why listening is always more important than looking. Turn your head away from the monitor or close your eyes and just listen before you move anything. Then repeat this same process after you move something to make sure you haven't made things worse.

Every beat doesn't have to be perfect. In fact, if it's too perfect, you'll suck the life out of the performance. Unless something jumps out as being really out of time, you might get away with leaving it as is. Another way is to just line up downbeats and any major accents, which gives you the best of both worlds: a natural feel that still sounds tight.

Copy and paste another section. If you have to make too many edits to a particular section, chances are it won't sound as good when you're finished as just finding a similar section from another part of the song and pasting it over the problem section. It's a lot faster and easier to do and will probably sound cleaner and groove better as well.

TIP: Many times the bass will speak better if it's a few milliseconds behind the kick drum rather than right with it. It will still sounds tight, but both the kick and bass will be more distinct, and the sound may even be fuller.

Listen against the drum tracks. If you listen to the track you're editing all by itself, you can be fooled into thinking its timing is correct when it's not, especially if you're editing to a grid. The real proof is when you listen against the drum tracks. If the mix element sounds great by itself and great with the drums, you're home free.

Trim the releases. This is one of the best things you can do to tighten up a track. Everyone is hip to tightening up the attacks so they're all on the beat or played with another mix element, but it's tweaking the releases that really makes the difference. Regardless of whether it's an accent played by the full band or mix elements, the song's ending, or a particular vocal or guitar phrase, make sure that the releases are pretty much the same length. If one is longer than the rest, trim it back and fade it so it sounds natural. If one is a lot shorter than the rest, use a time correction plugin to lengthen it a bit (see Figure 12.14).

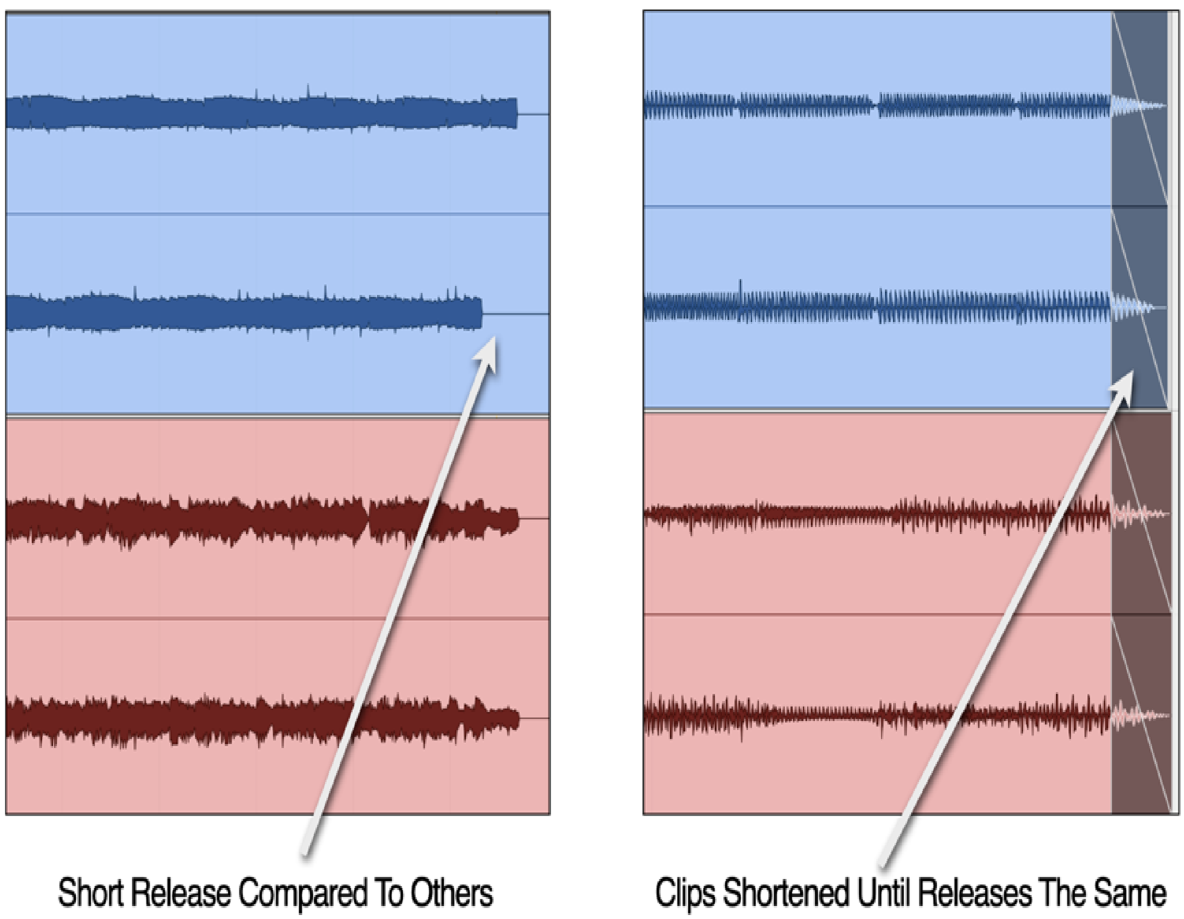


Figure 12.14: Timing of releases

Of course, if you're using loops, MIDI instruments or virtual instruments, you've probably quantized things to the track by now. If you haven't, remember that if it's perfectly synched to the grid, it may not sound natural.

The Secret To Smooth Song Endings

One issue I consistently come across in the song and mix critiques I do is uneven song endings. The majority of these have endings where the tail of the reverb or the fade just cuts off, or the fade on the last note simply isn't smooth. The biggest and most obvious problem here is that anyone who does listen to the end of the song will instantly categorize it as a demo and not a finished piece of work. Yes, really! Have you ever heard that on a major label release? Have you ever heard that after a song is mastered by a competent engineer? The answer to both these questions is a very loud "no".

So let's look at how to get those smooth song endings that you might have overlooked before.

You Can't Fade It If You Can't Hear It

First of all, if you're not putting a fade across all the mix elements on the end note, that's the place to start. While you can use a straight linear

fade, you might find that an exponential fade (see the graphic below) might actually sound better. Try both.

After that, either turn your monitors up or put on a set of headphones, because you want to be able to track the fade all the way to silence. The only way to hear that is by cranking up the level (don't worry, it won't hurt anything if you're just listening to the fade) or strapping on a set of phones on and cranking it there.

As soon as you get to silence after the fade, that's where you put your end point for your mix export. Congratulations, you now have a smooth and professional-sounding ending.

If you're sending your song out for mastering with a real engineer, the mastering engineer will do that for you and make any other fixes to the ending if needed. However, the only way for this to happen is if you provide enough of the decay tail for the engineer to do his or her stuff. Either way, follow this process every time and your mixes will be a lot better because they'll end a lot better.

The Ending Elephant In The Room

I just gave you the simple formula for a smooth ending, but it's still not going to sound as pro as you need it to be if you have instruments cutting off at different times during the fade. The ideal ending has all the mix elements holding that final note for as long as possible. If this is the case, your end fades will always sound smooth. This is what happens when you use well-seasoned studio musicians. They all know to hold that last note as long as possible, but most musicians that don't yet have

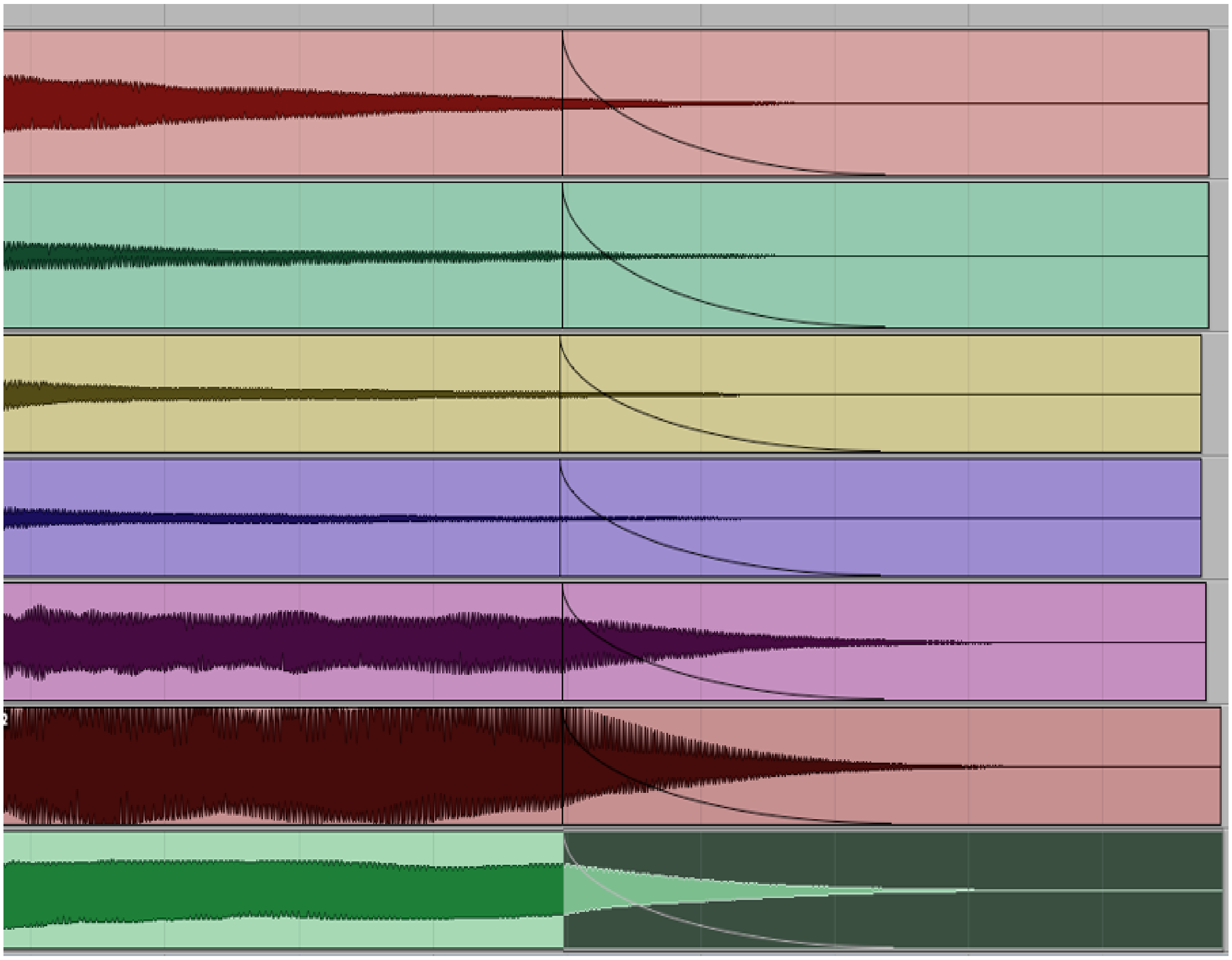
that studio savvy (or they haven't been beaten-up yet by an experienced producer) so they tend to cut off the last note too soon. In the mix, the result sounds disjointed even if you get the rest of the fade right.

So what do you do if the drums ring for three seconds, the guitar cuts off at half a second and the bass cuts out at one second? At that point, you have two choices: The easiest is to make sure everything cuts off with the shortest mix element, which in this case would be the guitar. We could cut the fade of the bass and drums off at 0.5 seconds, then make sure that the same fade is applied across all the other tracks. It could require a little experimentation to find the proper fade curve that makes this all sound right but you can see how it might look in a DAW timeline in illustration 12.15.

TIP: An exponential fade generally works well in this situation.

An alternative approach is to use time correction to elongate the last note of the guitar or any other short mix elements so they're the same length as the longer mix elements. Choose the correction method that sounds best, but make sure you pick one and execute it.

The problem with smooth song endings is that they're so often overlooked or worked on as an afterthought. They're really important if your goal is a professional sounding mix, though. Try the techniques outlined above and you'll be glad you did.



12.15: Smooth endings

Editing Songs With Playlists In Mind

Streaming has reshaped music in so many ways, but one area that's frequently overlooked is the change in the basic structure of a hit song. You may not be aware that hit songs these days are built quite differently than in the past, but they are. If you look at hits starting

from back in the 1950s up to around 2015, you'll see that the song structure was pretty similar and looked something like this:

Intro → verse → B section (sometimes called pre-chorus) → chorus → intro → verse → B section → chorus → bridge → chorus → chorus (fade)

There are fewer songs with this form anymore thanks to streaming. After analyzing dozens of songs on my What Makes This Song A Hit webinars for my Hit Makers Club members, these are the structural changes that are frequently seen today:

Songs start right in on the chorus.There's a good reason for this. Because of our short-attention-span mediascape, if you don't grab a listener right away you lose them forever. An artist can't expect a listener to wait 15 seconds before the lyrics begin and a minute or longer for the song's chorus, because you'll probably lose that listener to some other stimulus, especially if they're not already familiar with the song. Hence, hits today get right to the point as soon as possible.

There are no bridges.The bridge was traditionally the peak development of the song, and although the choruses afterwards maintained a similar high intensity, there was usually a fade at the end instead of a hard ending. Today's hits still have plenty of development, but the peak usually comes in the last chorus rather than a bridge.

There are no song fades. Most hits today have hard endings with some being more eloquently designed than others (some just do feel like an abrupt stop because nothing else was recorded). Actually there's no need for a fade today. The long intro and the fade-out were primarily designed

for radio DJs to talk over, and since today's artists care little about radio play, why insert a section that is effectively obsolete?

There are no solos.You'll be hard pressed to find anything that even comes close to a traditional "solo" these days. Artists, producers and musicians aren't concerned about whipping a solo out because there's no longer a need to lengthen what might once have been considered a too-short song.

Songs are shorter than ever. When it comes to streaming, you get paid the same regardless of whether a song is 1:40 or 4:40 so why not make it short? Maybe the listener will play it a second time and you'll get paid twice, as opposed to listening to a longer version with its protracted fade-out and risk having no desire to hear it again. Short-attention-span meets leave-them-wanting-more.

Songs are somewhat homogenized thanks to so many collaborators. It's rare to find a single songwriter or even a duo writing team on a song these days. You're more likely to find ten or more writing credits, and perhaps that many on the production team as well. Some of this can be attributed to the "cover your butt" mentality of avoiding lawsuits down the line, but much of it is simply corporate in nature in the sense that creating a commercial hit is a more important consideration than creating a great song. That's not exactly the way to set a new music trend or change people's perceptions of ourselves and the world around us…but that's not the nature of songwriting today.

Keep in mind that this change in song structure doesn't necessarily lead to "bad" music (that's so subjective anyway) and commercial gain has always been a driving factor in the creation of music. It's called the "Music Business" for a reason. But it's still worth understanding how commercial forces — as well as technical forces — continue to change

the way music is created and sold. Like everything else, music creation and music tastes change with the times. It’s not good or bad, it’s just a fact.

2 Tips For Editing Songs For Streaming

As stated earlier, many songs are now written primarily for streaming distribution but this doesn’t mean you shouldn’t stand by your creative impulses and take a song wherever it needs to go. It just means that you might have to consider a bit of editing later while keeping the nature of playlists in mind.

The idea of editing a five minute song down to three minutes or less is inherently repulsive to many artists but, don’t forget, this has been done for decades with songs being shortened to fit broadcast radio formats since the 1950s. Yes, we hated these edits if we were already familiar with the full song, but they got the job done by exposing the song to a larger audience, which is all you’re trying to do here.

Another reason for a playlist edit is that it provides something every artist needs: more material to release. Your fans love it when you offer them different versions, and every release is a new event to promote. The longer, original version of a song currently in circulation clearly fits the bill.

So what kind of edits are required to make a song more viable for a popular playlist?

1. Start the song with the main hook or chorus. The idea is to keep the listener from skipping to the next song. The goal is to have them hang in with you for at least 30 seconds, which is long enough so you get credit for a listen. If the song has a long intro, edit it down to 10 seconds or less, or eliminate altogether. The hook is everything in streaming.

2. Cut out the fat. You may have played a great guitar solo but it may be adding an unnecessary (at least in this case) 30 seconds to the song that might dissuade someone listening to the end. Cut it out, as well as any long intros, outros or fades.

Did you hack up your song? Yes, you've probably made it significantly shorter. Are you happy with it? You might be surprised and find that the shorter version is more to the point and flows better.

That said, remember that the idea here is to expose more people to you and the song, and by making it playlist-ready you're doing just that. And don't worry – people will still hear the full version of the song when you release that too. The challenge we all face is for them to know your work enough to want to hear more of it.

Pitch Correction

Depending upon how much of a purist you are, pitch correction is either the worst thing to ever happen or a godsend. Regardless of how you land on the issue, pitch correction is, at the very least, a necessary evil in today's music.

Tuning vocals has been done since the early 1970s, starting with the first Eventide H910 Harmonizer (see Figure 12.16). Primitive as it was, it did allow for slight pitch changes, although the digital artifacts it imposed on the audio were quite substantial.

With each new model of pitch shift hardware that was subsequently introduced, the technology improved to the point where today there are some excellent plugins commonly available that would simply astound any engineer transported in time from back then.

Figure 12.16: An Eventide H910 Harmonizer

Courtesy Eventide

There are three popular track-tuning programs commonly in use: Antares Auto-Tune, Waves Waves-Tune, and Celemony Melodyne, as well as less-often used variations, such as Avid's Elastic Audio or Cubase VariAudio. Be aware that all tuning programs impart their own "personality" on the audio you're tuning, and it might not always be pleasing, which is why many engineers own several different versions so they can compare which sounds better in a particular situation.

Although the manner in which pitch correction plugins are used is more or less the same, there are a number of guidelines worth following to keep the mix element sounding natural:

Use the performance itself first. Before you apply pitch correction, exhaust all other remedies to keep the performance as unaffected as possible. These include comping together multiple takes and copy-and-pasting phrases, words, or even syllables from other parts of the recording.

A little goes a long way. The fewer notes you correct, the more natural the performance will sound. You're much better off just correcting a specific few notes here and there than attempting to correct an entire performance.

TIP: Generally, background vocals can get away with much more pitch correction without being obvious than lead vocals.

Use the most precise mode. Many engineers avoid using Auto mode because it's not precise enough for most applications and as a result, causes audible fluctuations that make a voice sound noticeably auto-tuned — which is usually not what you're after (Cher and T-Pain aside). If your plugin has a graphical mode, use that to achieve the most precise tuning with the least amount of artifacts.

Don't perfectly tune the vocals. Even the best vocalists are never precisely on pitch so if you tune it that way, it may sound unnatural. Getting the pitch within a few cents will sound more like the real thing,

since it's the variations and inaccuracies that make a human voice sound human.

"Of the four albums I co-produced with David Bowie, 95 percent of all his vocals were only one take. There are small errors in pitch and timing if you listen closely, but that's what makes David sound like David."

—Ken Scott

Print the pitch correction. Instead of leaving the pitch correction patched in as a plugin, you're better off if you print a corrected pass and use that vocal instead. This not only saves precious system resources, it also eliminates any problems that might occur if the session is moved to a DAW with a different software version or the plugin has a broken update down the road.

Pitch Correction Techniques

Here are a few techniques often used when correcting the pitch of a mix element. As always, don't be afraid to experiment, since slight variations might fit better on a particular mix.

Sometimes raising the formants (the harmonic resonance) of a voice track can make the vocal sound a bit more exciting or breathier. It doesn't actually change the pitch, just the placement of the voice's harmonics.

If the vocal is off enough that the pitch correction sounds "alien," try this trick from the old Harmonizer days: copy the vocal to two additional tracks and spread them left and right. Tune one of the vocals up by 2 to 8 cents; tune the other down by 2 to 8 cents. This will smooth an out-of-tune vocal while at the same time make it sound a lot thicker.

Tuned vocals almost always sound better with a least a touch of reverb or delay on them.

Sound Replacement

Although you may be great at recording drums and have a great sounding studio with an excellent signal chain, the two chief variables in the recording are the drummer and his drums. No amount of technique or gear can overcome a bad sounding kit or a drummer that hits inconsistently, hence the importance of sound replacement software and a good sample library.

Drum replacement is a technique pioneered by the late, great engineer Roger Nichols during the 1970s while working with Steely Dan. In the days before DAWs, Roger developed his own hardware device called the Wendel that was an ultra-fast set of triggers meant to replace real drum sounds with more consistent and better sounding ones from his library.

Today there are numerous drum replacement plugins that do the job equally well and it seems like every mixer has his or her own preference, be it Drumagog, Steven Slate's Trigger, Superior Drummer, or any of

the other plugins that may be included with a popular DAW software package. Modern drum sound libraries can include those that come with the software, third-party libraries you can purchase a la carte, and personally made libraries consisting of the engineer's own samples.

While the original approach was often to completely replace a sub-par drum track, many mixers now only do this as a last resort, opting instead to double the original sound with a sample to help keep the human feel and the basic drum elements intact. The new sampled drum acts as an acoustic EQ, changing the sound by augmentation instead of electronically changing the frequency bandwidth of the originally recorded drum. Although some mixers try to delete the leakage between the various drums of the kit, that leakage is what gives a kit its particular sound, and doubling the original kit with a sample helps to maintain some of that relationship.

Keeping The Sound Natural

It's very easy to make a drum track seem more like a drum machine than a real human if you're not careful. For drum replacement to sound natural, the following must occur:

The sample should sound better than the drum it's replacing.

The sample should blend seamlessly with the other drums in the kit.

No one should be able to tell that you've replaced the sound.

To replace drums, it's important to understand that any drum sound is composed of two different parts: the initial transient hit, and the resonant sustain and decay that makes up the body of the sound. It's much easier to trigger a replacement drum if the initial transient hit is strong, since it will be clear to the software exactly when to trigger the sample.

TIP: Sometimes it's easier and more natural sounding to replace a sound manually by copying and pasting a hit or fill from another place in the song, especially if there are only a few hits that need replacing.

It's usually best to record each replacement sample to a new track rather than to keep the drum replacement plugin inserted as a plugin during the entire mix. This saves system resources (some sound replacement plugins are resource hogs) and makes your mix more transportable because the sounds are printed and therefore locked-in.

Sound Replacement Techniques

There are many techniques when it comes to sound replacement or enhancement. While most refer to drums, it's possible to use some of the same techniques in other situations as well if you're willing to experiment.

If the original drum sounds okay soloed but disappears in the track, select a sample with a high initial transient.

If the original tone of the drum doesn’t fit the song, select a sample based on the character of the body of the sound after the initial hit.

The simpler the drum pattern, the easier it is for a sound replacement plugin to recognize the drum hits and trigger the sample at the right time.

Most drums become brighter the harder they’re hit; so choose a sample that features these characteristics in order for it to sound natural.

If the drummer plays the snare using mostly a sidestick, be sure that the sample has mostly sidestick as well.Timing is extremely important when adding a sample to the original sound so take care to ensure that both transients line up together. Many sound replacement plugins have some latency, so be sure to check this closely and move the sample as necessary.

The easiest way to find the transient is to use the Tab to Transient feature in Pro Tools, Hit Detection in Nuendo, or any other transient detection feature that your DAW has to offer. As an alternative, you can use any available Strip Silence–like feature on a copy of the original track to make the transients more distinct.

Well-played drums with a few ghost notes will beat robotic-sounding, replaced drums every time.

It's critical that you check the phase alignment of the sample against the original drum. Even though it may look to be in phase, your ears may tell you something completely different when the Phase button is selected. Snare drum is the hardest to augment/replace because of the nuances in how hard and how frequently the drummer hits it. Be sure to use a multi-sampled snare drum with multiple level variations to keep it sounding like it was played by a real drummer.

If you don't have a multi-sampled snare or the sound doesn't have enough sample levels, you can make your own by duplicating the sample a few times and raising and lowering the pitch of each one by 2 cents. Slightly changing the pitch gives the impression that the snare is hit a bit louder or softer.

If the drummer plays a lot of ghost notes on the snare, sometimes it's best to make a copy of them and concentrate on replacing only those. This allows you to fine-tune the sound replacement software only for ghost notes.

Many times the best way to trigger the snare drum is by using the sound from the under-snare mic if one has been recorded. It's usually a highly transient signal that can be easily recognized by the sound replacement plugin.

Sometimes using a gate on the original drum track can maintain just the transient hit, making the sample that much easier to trigger. This works

especially well if you're trying to replace the body sound of the drum rather than the hit part.

Sometimes compression of the drum track will provide a more consistent drum trigger.

Many engineers always have the drummer give them solo hits of each drum while recording so they can use them as samples later.

TIP: You can make your own samples for a specific song by taking the best sounding hits from the original recording of the kit you're mixing and triggering them as needed.

Automation

Automation is the process of recording parameter adjustments so your DAW software can automatically execute them during playback. Before automation, any mixing moves during a complex song had to be made by the engineer, the assistant, the band members, and/or anyone else who had their arms near the console. While this might have been a very "organic" way of mixing, it was not repeatable in the slightest, so engineers everywhere longed for a way their fader moves could be memorized and played back later with some degree of precision.

The first primitive console automation arrived during the 1970s and was known as VCA automation because it evolved around a component

called a Voltage-Controlled Amplifier. This method commandeered two tracks of the tape recorder to store the automation data and any updates, and while it remembered your fader moves, the faders themselves didn’t actually move. This was obviously confusing and became really unwieldy as consoles became larger and larger. Also, having the VCAs in the signal path degraded the sound as well. The system also suffered from increased latency with each successive update.

The next generation of console automation system actually provided moving faders (the first one was introduced by Neve and called NECAM), which were extremely expensive, as was the corresponding computer that memorized their movements. Don’t forget, this was in the pioneering days of computer science, when even 500 kilobytes of RAM (yes, kilobytes, not the far larger gigabytes that we’re used to today) cost thousands of dollars.

Modern mixing has always been a complex and dynamic operation where a long list of console parameters are set during each mix, so it’s important to remember each parameter’s exact setting in order to make a remix of the song at a later date.

The first generation of console automation to make that possible was SSL’s Total Recall in 1981, which took a snapshot of all the switches and parameter controls on the desk. The problem was that the reset had to be done manually, usually by an assistant, which could take two or three hours to complete on a console with 48 or more channels.

For a brief period, resettable consoles became all the rage, where most of the console parameters would reset themselves automatically without the help of a human. Although this evolution began during the analog age, it became much easier and cheaper to implement as digital

technology found its footing and is now pretty much standard in most professional consoles, even the relatively inexpensive ones.

But now we're in the age of the DAW, where dynamic automation of virtually every parameter is the norm. This allows mixes to be more intricate than ever and take more time than ever as a result, but the pinpoint accuracy of every parameter movement during every millisecond of a mix is assured.

Fader Automation

Most automation systems, regardless of whether they're in hardware on a console or in a DAW, have the following modes:

Write. Used the first time the automation data of a channel is recorded or when writing over the existing automation data.

Touch. Writes automation data only while a fader is touched. The fader then returns to its previously automated position after it's been released.

Latch. Starts writing automation data when a fader is touched and keeps recording the automation after it's released.

Read. The written automation moves are replayed.

Off. The automation is disabled.

Besides the fader moves, all of the above modes include the channel mute button at the very least, and they typically extend to other parameters such as panning, aux send levels, insert in/out, and other parameters as well on most DAWs.

Generally speaking, most mixes are begun with the automation off until the basic balance is complete and any signal processing has been inserted. The first automated pass will be in the Write mode to initially record the static balance and parameter settings. From that point, most automation is written in Touch mode, where the changes are written as long as the parameter is being changed but then return to their initial state after the changes are completed. Occasionally a track may be written in Latch mode, where a brief change is required but it must then carry on recording automation until the end of the track.

It's not uncommon for a track to have many automation moves but then require that the level of the entire track, including all the already captured moves, be either raised or lowered. Most automation systems have what's known as a Trim control that allows the level of the entire track, or sometimes just a portion of it, to be changed by a selected amount. Another way to accomplish the same thing is with a level Trim plugin (see Figure 12.17).

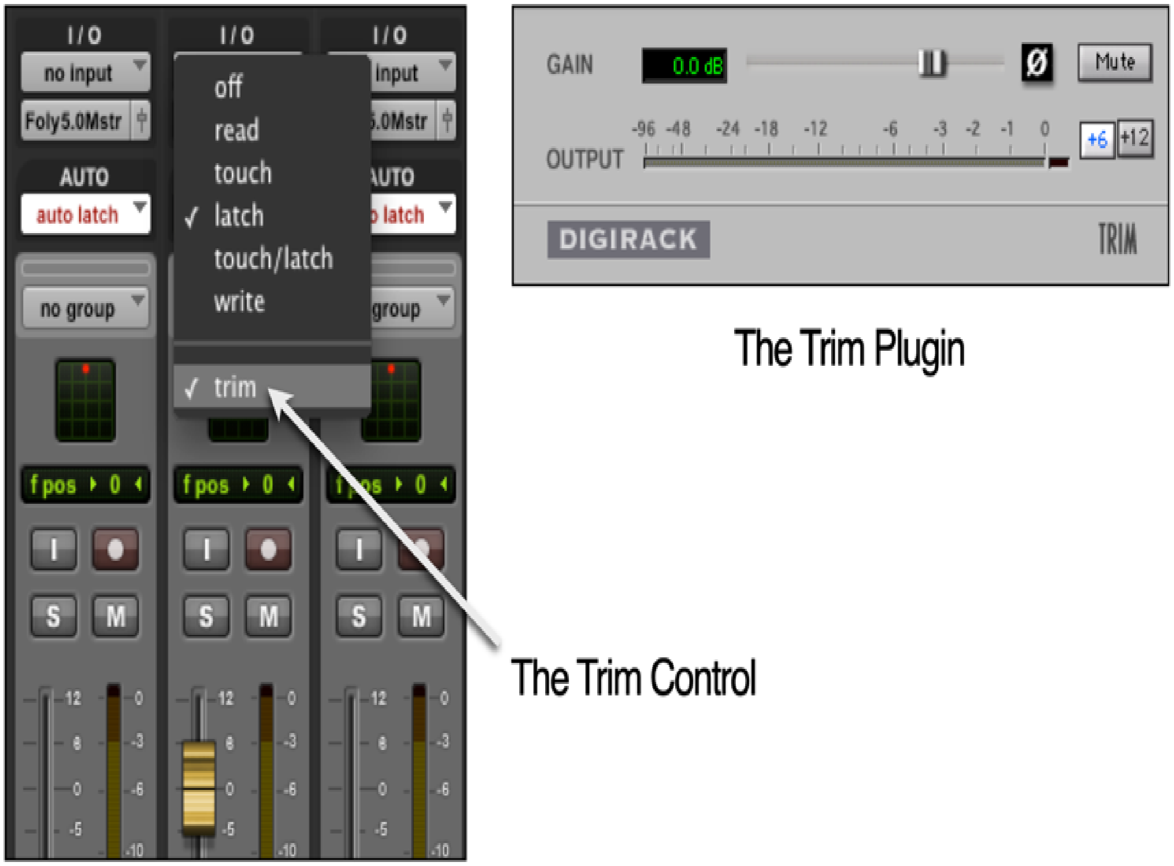


Figure 12.17: The automation Trim control

Drawing The Automation

In many cases, it's much easier to draw the automation in than to use a fader, especially if you need a higher level of precision. This can work if you know the exact curve you want the automation to follow but sometimes it's easier to adjust the curve of what was previously

recorded as fader automation, perhaps to trim the move shorter or longer (see Figure 12.18).

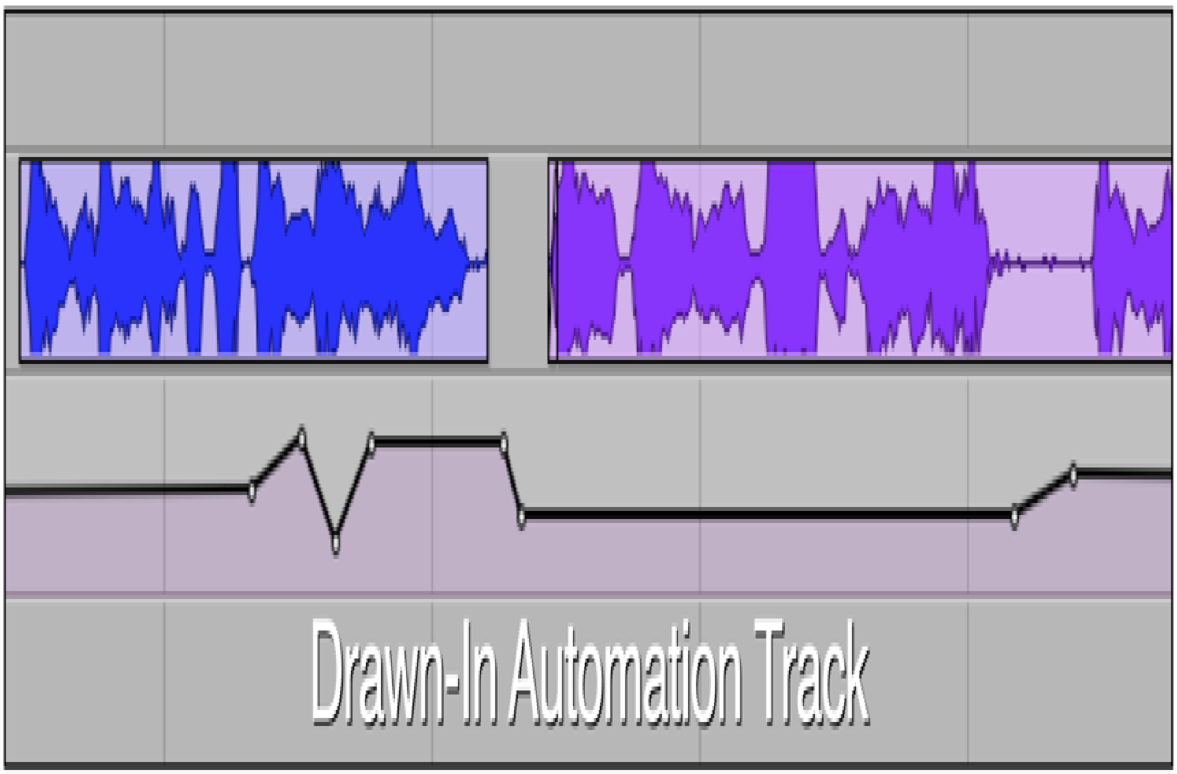


Figure 12.18: Drawing the automation

Using Automation To Add Dynamics

For the most part, a mix where the faders remain more or less static can be boring and unexciting. Even before automation, mixers were constantly riding instrument and vocal faders during a mix to make sure they stood out in certain places or added an extra intensity to the mix. The best part about automation is that those moves can be exactly replicated on every pass.

TIP: The key to understanding how to use automation to add dynamics is by observing a performance by a great band. This will help you be able to hear all the nuances that the dynamics of the mix need for it to be exciting.

Among the ways to add dynamics to a mix are:

Slightly boost the rhythm section during fills, turnarounds, and even choruses (usually only a couple of dB is all that's necessary but this depends upon the track).

Boost the snare and toms during fills.

Boost the kick, snare, or cymbals on accents or the downbeat of a new section.

Duck the rhythm instruments during an instrument solo to help clear out space in the mix.

Boost the hi-hat in parts where it’s being struck and decrease it where it’s not.

Add additional reverb or delay to an instrument when it gets masked as other instruments are added to the mix.

Pump a strumming rhythm guitar in time with the music, pushing it especially on 2 and 4, or push it on the upbeats (one AND two AND three AND…).

Gently boost the fills or other instruments between vocal phrases. Pull back the downbeat of a chorus if the drummer hits it too hard. Pump the “4 AND” on a percussion track.

Automation Techniques

Each of these techniques may apply to applications other than the one specified so keep an open mind, especially when boosting the dynamics.

Many old-school engineers use fader automation to control the dynamics of a vocal rather than use a compressor. This is accomplished by both riding the level of each phrase (or word or syllable) and riding the end of vocal phrases.

Ride the bass to even out the energy differences of certain notes.

Ride an instrument's sustain at the end of solos.

Ride the reverb returns during parts where it gets lost.

Use the Trim tool to adjust the automation of a track up or down slightly, if necessary. No need to redo the track if the vibe is okay but the level isn't.

Another way to adjust the automated track is to adjust the output volume of the last inserted plugin up or down as needed. If the track doesn't have any plugins, insert any one that has the ability to control its output without actually inserting the effect.

Template Mixing

An intriguing way some mixers work is by creating a template, which is a preset DAW configuration that they can simply plug audio tracks into and quickly start mixing. Most mixers have at least a basic setup, which might include effects, subgroups, and plugins that are almost always used. In fact, some mixers have multiple templates to use as starting points, each for a different type of music or genre.

A more advanced template strategy uses a template with channels dedicated to each mix element with plugins and even compression and EQ settings all preset. In other words, the kick always goes into the same channel of the preset, the snare to its own, the bass to its own, and so forth.

Mixer Chris Lord-Alge is credited with being the first to take this approach back in the analog era of the 1990s, centered around an SSL console. Nashville mixer Billy Decker brought the same approach to the digital era using a DAW (you can read how he does it Chapter 18). Both mixers claim that the time to mix a song has been reduced to less than an hour. The fact that both mixers have mixed huge hits shows that template-based mixing can really work. Both have won Grammys, with Chris mixing hits for Madonna, Bruce Springsteen, Prince and James Brown, while Billy mixed number one hits for Sam Hunt, Chris Young, Rodney Atkins, Dustin Lynch, Parmalee, Colt Ford, and many others.

What makes template mixing work? One theory holds that it's because the level of each track hits the compressors in just the right way, then the other processors and effects fire from there. With Billy, the idea is to make sure that the waveform fills about half of the DAW track height. All processors are set to work best at that level.

If you're new to mixing, you should probably learn to mix by following the guidance of this book as well as its companion "The Music Mixing Workbook" before you experiment with any advanced techniques. That will give you a good foundation in mixing fundamentals first. After that, template mixing is something worth checking out.

The Master Mix

Gone are the days of manual mixing, the days when the nimble fingers of not only the mixing engineer but also the studio assistant, the producer, he producer's assistant, someone's friend, and one or all of the band members manned a fader or a mute button or a pan control to get the perfect mix. Gone are the days of an endless number of live takes of the mix to finally get your "keeper." Thanks to digital workstations and the advanced state of console automation, if you work in a studio, the mix can be perfect before it ever gets committed to a hard drive, analog tape, solid-state memory, or any other format yet to be devised.

That said, how do you know you've reached that point of mix perfection? That place where it just won't get any better? And when your mix has reached that point, what do you do with it then?

These are the questions this chapter will answer.

Eight Indicators That Your Mix Is Finished

One of the toughest decisions you'll face as a mixer is knowing when the mix is finished. If you have a deadline, the decision is made for you as the clock ticks down, but if you have unlimited time or deep-enough pockets, a mix can drag on forever.

So when is a mix considered finished? Here are some guidelines:

The groove of the song is solid. The song's pulse is strong and undeniable.

You can distinctly hear every mix element. Although some mix elements (such as pads) are intended to blend seamlessly into the track, most mix elements should be clearly differentiated.

Every lyric and every note of every line or solo can be heard. You don't want a single note buried. Everything has to be crystal clear. Use your automation and clip gain. That's what it was made for.

The mix has punch. The relationship between the bass and drums (even if they're samples or loops) is in the right proportion and works well together to give the song a solid foundation.

The mix has a focal point. What's the most important element of the song? Make sure it's obvious to the listener.

The mix has contrast. If you have too much of the same effect on everything, the mix can sound washed out. Likewise, if your mix has the same intensity throughout, it can be boring to the listener. To keep the mix interesting you need to build contrast between its elements: from dry to wet, from serene to intense, from complex to lean.

All noises and glitches are eliminated. This includes any count-offs, singer's breaths that seem out of place or predominant because of vocal compression, amp noise on guitar tracks before and after the guitar part has played, bad-sounding edits, and anything else that might take the listener's attention away from the track.

You can play your mix against songs that you love, and it holds up. This is perhaps the ultimate test. If you can get your mix in the same ballpark as many of your favorites (either songs you've mixed yourself or mixes from other artists) after you've checked-off the previous seven items, then you're probably home free.

How much time should all this take? In the end, most mixing pros figure at least a full eight-to-twelve hour day per song regardless of whether you're mixing in the box or mixing on an analog console. While a complex mix may take several days or more, it's still best to figure a day and a half per mix if you're mixing in a studio with an analog-style console and traditional hardware outboard gear. Then again, Billy Decker can do it in 45 minutes with his template mixing technique. If you're mixing every session in your DAW as you go along during recording, then you might be finished before you know it, especially since all you may have to do is tweak your mix a little here and there to call it complete.

Competitive Level

Since as far back as the 1950s, mixers have strived to make their mixes hotter than their competitors'. The reason for this is because when two songs are played back to back, the louder one is often perceived to sound "better." Of course, the limitation of how loud a mix could

actually be was determined by the delivery medium used to get the music to the consumer. In the days of vinyl records, if a mix was too loud or had too much bass, the stylus would vibrate so much that the record would skip. When mixing too hot to analog tape, the sound would begin to distort and the high frequencies would start to disappear. When digital audio and CDs came along, any attempt to mix beyond 0dB full scale would result in horrendously bad-sounding distortion as the digital signal was clipped.

Even with the built-in limitation of 0dBFS, over the years mixes have become hotter and hotter in perceived level, mostly because of a new digital technology that resulted in better and better limiters. Today's digital "look ahead" limiters make it easy to set a maximum level (usually at -0.1 to -1dB FS) and never worry about digital overs or distortion again.

That being said, raising the competitive level (a mix level that's as loud as your competitors' mixes) used to be left to the mastering engineer. The mix engineer would hand off a mix that was deemed acceptable by all parties involved (artist, producer, management, A&R, etc.), and the level would then be raised to what was appropriate for the medium from there, regardless of whether the ultimate delivery medium to the consumer was a vinyl record, a cassette, CD, or Internet stream. Part of the voodoo of the mastering engineer was his or her ability to make your mix sound louder than you, the mixer, were able to achieve in the studio.

Mixing like that doesn't cut it any more. Artists, producers and record execs want the mix to immediately sound not only "like a record," but as loud as anything commercially released from the first rough mix onward. This is one of the reasons why the mix buss compressor on all SSL consoles became so popular. It was built like a typical mastering

compressor to give your mix that "radio" sound as soon as you inserted it in the signal path.

Today, with the many powerful plugin compressor/limiters available, it's all too easy to raise the level of your mix as loud as it will go, but just because you can do this doesn't mean you should.

Tips For Hot Levels

If you're mixing your own material this might not be important, but if you're mixing for someone else, it's important to attain relatively hot levels on mixes so they can roughly compare them to existing recordings. Failure to do this will probably get a, "Why doesn't it sound as loud as. . .?" response, and no amount of explanation will take the place of just giving them a mix that's hotter. Here are some tips used by mastering engineers that you can use to do just that:

Set a compressor at a 1.5:1 or 2:1 ratio with a slow attack and release.

Set the compressor for up to 5dB or less of compression.

Raise the output of the compressor to the desired level.

Feed the compressor into a modern digital limiter that has a Ceiling control and set to a maximum of -0.1 to -1dBFS to prevent digital overs.

Tweak the gain of the limiter to taste.

DON’T HYPERCOMPRESS.

Figure 13.1: Master buss signal path for hot levels

The real key is to use a compressor with a low compression ratio (1.5 or 2 to 1) and a slow attack and release time across the mix buss. This provides what's known as relative level, which means that it increases the perceived loudness of the track. The compressor then feeds a look-ahead-style limiter set to clamp down at -0.1 to -1dBFS to prevent digital overs. Additional limiting is then used to taste (be careful—not too much!).

TIP: Ask the mastering engineer what he or she needs before sending a master. Remember, if your client signed off on a mix, be cautious on changing anything even for mastering!

Many times an EQ is also inserted between the compressor and limiter to provide some girth at 60Hz and some air at 10kHz. This is usually a rather clean plugin like a Sonnox Oxford EQ or FabFilter Pro-Q2 or 3, but anything that gets the sound you want will work. Usually just a couple dB in each band is all that's needed. If you feel you have to massively equalize the mix buss, you might be better off examining your mix for the source of the problems.

Hypercompression

Too much buss compression or over-limiting either when mixing or mastering results in what has become known as hypercompression, where the waveform is cut off at the top or "flat-lined" (see Figure 13.2D). Hypercompression is to be avoided at all costs because:

It can’t be undone later.

It can suck the life out of a song, making it sound weaker instead of punchier.

Online codecs have a hard time encoding hypercompressed material and may insert unwanted sonic side effects as a result.

It leaves the mastering engineer no room to work, so his or her efforts are less effective.

Many online music services now normalize the uploaded content, so being too loud has no advantage and can even sound worse than a song with a lot of dynamic range.

A hypercompressed track has little dynamic range, leaving it loud but lifeless and unexciting. On a DAW, it’s a constant waveform that fills up the DAW timeline region. As our tools and tastes have changed over the years, here’s how the levels on recordings have changed (Figure 13.2).

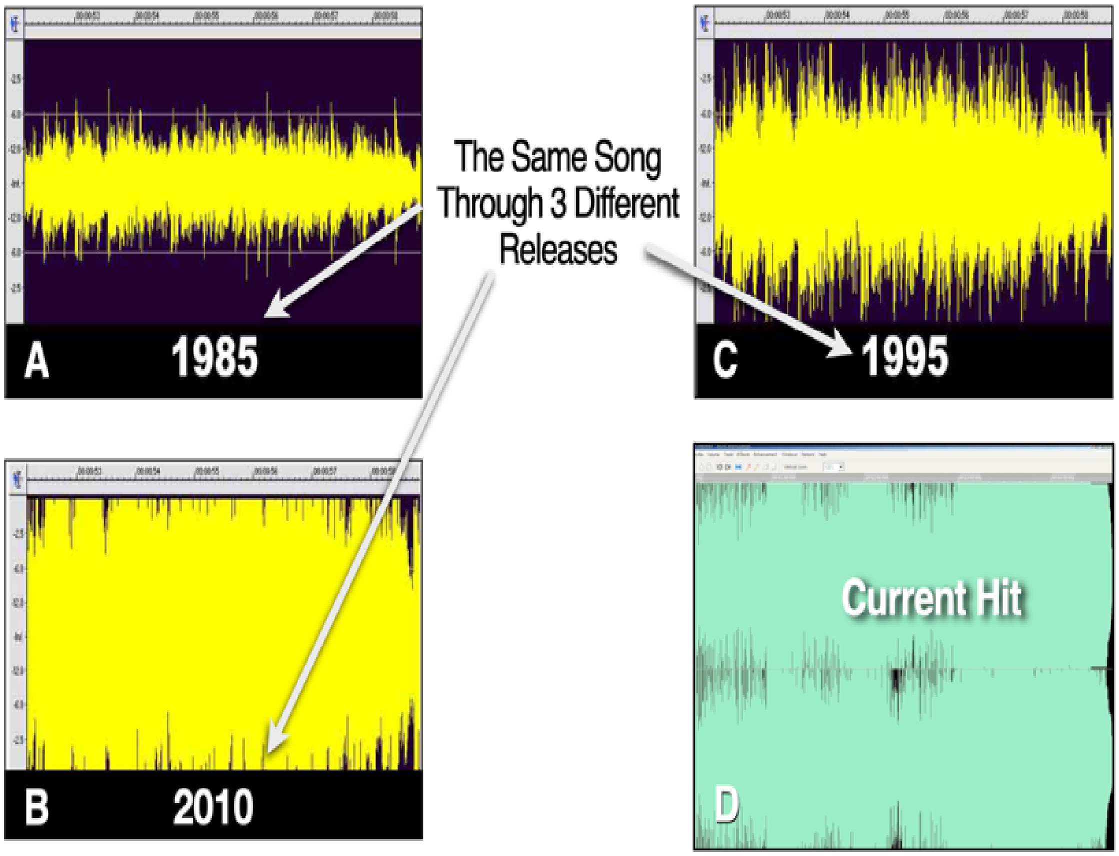


Figure 13.2: The evolution of hypercompression

Although everyone wants a hot mix, making it too hot might soon become a thing of the past in this online world that we now live. As you'll soon see, online music services such as Apple Music and Spotify now normalize the uploaded content so it all plays at exactly the same level regardless of how hot or quiet the original uploaded master played.

As a result, there's no longer a good reason to make the master levels extremely hot, and it fact, it's proven to actually be counterproductive. A song with more dynamic range and less level will actually end up sounding better than the louder one after the service's normalization process. Mastering engineers everywhere (as well as mixing engineers) will jump for joy as sanity returns to mixing levels and the "volume wars" finally end. We're not quite to that point yet, but for the first time in many years, it looks like progress is being made.

Using LUFS For Mixing

You might have noticed that in the last few years the differences in level between television shows, commercials, and channels are pretty even, with no big jumps in volume between various programs. That's because viewers were complaining about the dramatic increase in level whenever a commercial aired because it was so compressed compared to the audio of the program you were watching. Congress set out to do something about this, and in 2012 adopted a method to normalize those volume jumps that the European Broadcast Union put into place the year before – Loudness Unit Full Scale or LUFS.

LUFS (called LKFS in Europe) is a way to measure the perceived loudness of a program by measuring both the transient peaks and the steady-state program level over time using a specially created algorithm. It's different from a normal meter in that it doesn't represent signal level at the moment – it measures how loud we perceive an entire audio program to be. For a broadcaster this is actually pretty serious, because if a station violates the mandated LUFS level of -24 in the United States, it could potentially lose its broadcast license.

Even though LUFS was intended primarily for broadcast audio delivery, it has a new increased meaning in music production as well. Thanks to the fact that streaming services like Spotify and Apple Music are now normalizing the songs so the level is the same from tune to tune, there's no real benefit in compressing a song to within an inch of its life any more. In fact, less volume and more dynamic range are actually your friend.

Using a LUFS meter across you mix buss allows you to optimize your music mixes for a variety of platforms to be sure that they're always in the sweet spot for dynamic range. There are a number of LUFS meters on the market from TC Electronic, Waves or other developers, but one plugin in particular from Mastering The Mix called LEVELS (see Figure 13.3) shows you not only a safe dynamic range for your mix, but also allows you to zero in on the level of the bass bands to make sure the mix has just the right amount of low frequency energy.

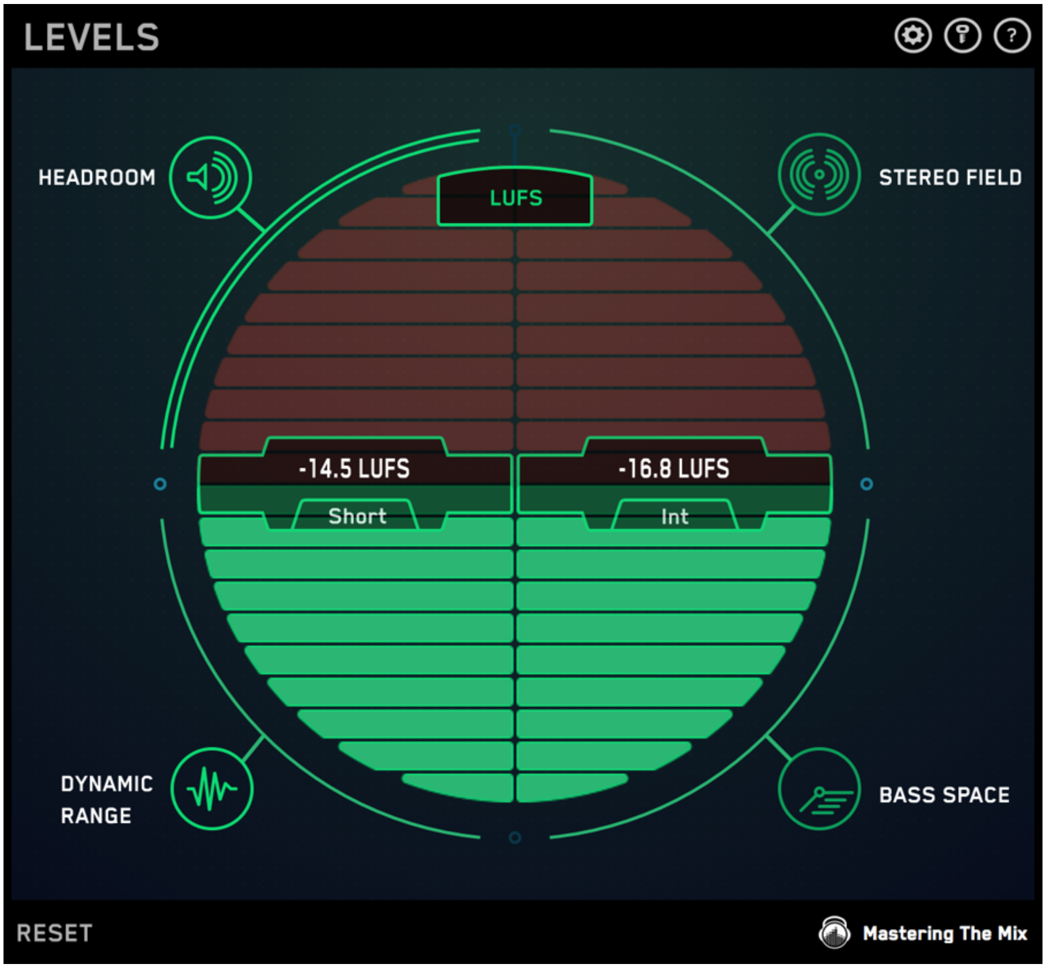


Figure 13.3: LUFS mixing with LEVELS plugin from Mastering The Mix

Courtesy Mastering The Mix

Standard LUFS Levels

Looking back at the analog days, mixing level requirements seemed so easy. You aimed for 0 on the VU meter and didn't worry too much if it bounced over. Of course, under the hood, 0VU could actually be calibrated to different levels but we usually didn't concern ourselves too

much with that as long as it was clean around the 0 mark. These days there are so many different meter reference calibrations available that it can take some time to settle on one that you feel comfortable with. That said, LUFS looms large when it comes to mix delivery signal levels, and that makes for lots of confusion.

Most audio metering looks at what's happening right now in the moment, while LUFS can do that with its Short Term setting, the Long Term and Integrated settings that are most often used look at a more or less average level across the entire program, plus it incorporates a smart algorithm that matches how we hear.

LUFS has worked so well in television that other parts of the audio industry adopted it and that's where the confusion sets in. There are a lot of different standards and that can mean confusion for the average music mixer. Let's try to sort this out as we look at some standards.

-24LUFS. This is your target if you're mixing for television (-23LKFS in the rest of the world). Out of all the standards, this is the most serious because a television network can get its broadcast license revoked for a violation. Send in a program with a higher level and it will be kicked back for a revision to the correct limit.

-16LUFS for gaming. This isn't as hard and fast a regulation as with broadcast television but it's what the gaming development community has settled on as a standard. Nothing serious will happen if you violate this level (you won't lose your game developers license) so it's okay if it's a little off.

-16LUFS for podcasts. Since Apple iTunes has been the principle delivery system for podcasts since the beginning of that format, this is the level it settled on and the rest of the industry happily followed along. As with gaming, it's not a big deal if you're off a little.

-16LUFS - Apple Music (also the AES standard). You don't have to deliver this level as they're going to re-encode it anyway

-14LUFS - Spotify, YouTube Music, Tidal. You don't have to deliver this level as they're going to re-encode it anyway.

-9LUFS or less for music. There is actually no standard LUFS level for music, but it's been determined that songs mixed to this level are sufficiently loud yet still contain some dynamic range.

LUFS In Music

Right now, you're probably saying, "…But Apple Music's standard is at -16LUFS and Spotify is at -14 and YouTube is at -14 and my favorite mastering plugin says -9! What is going on here?…"

Here's the thing, no matter what LUFS level you mix master is at, Apple, Spotify, Tidal, Deezer and every other streaming service will normalize it to the standard of their service anyway. You're just making more unnecessary work for yourself as a mixer by mixing to different levels for different streaming services (see Figure 13.4)

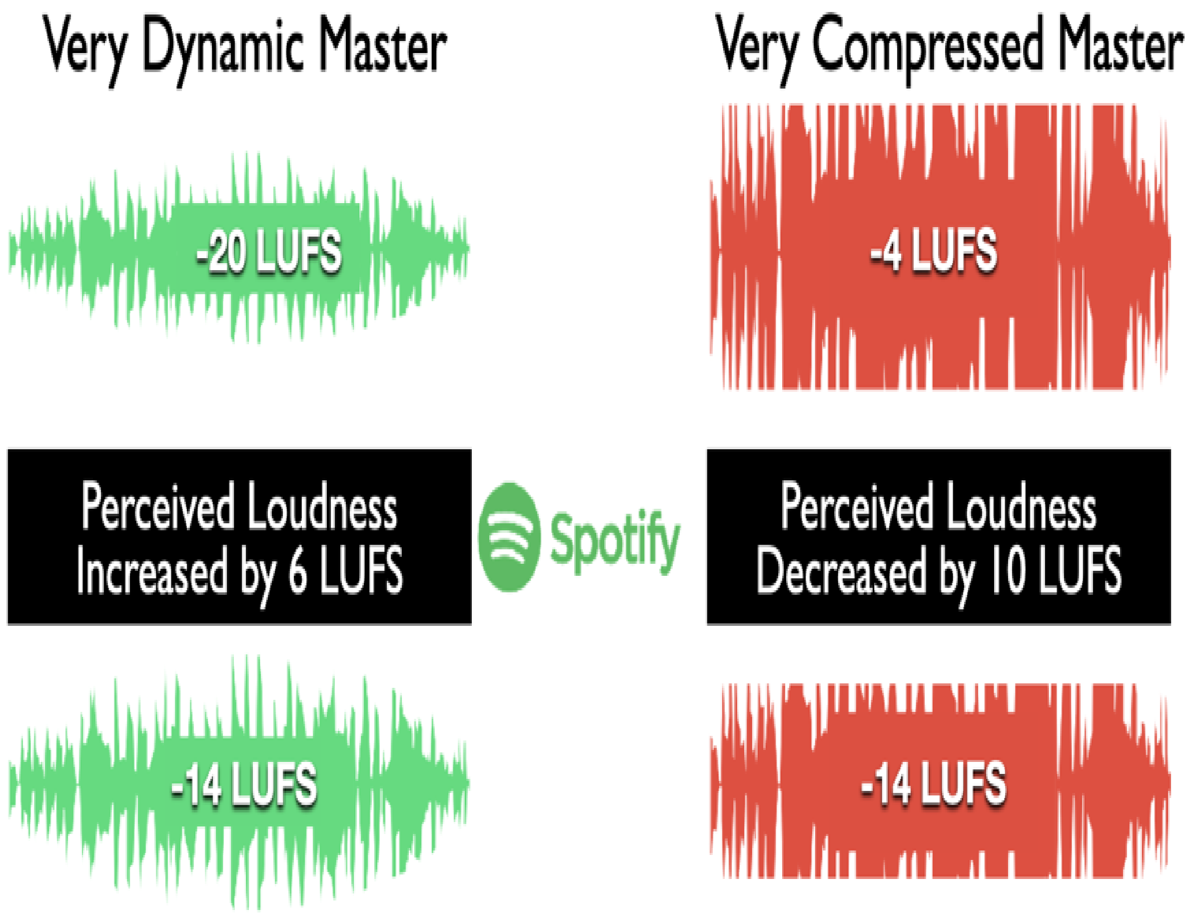


Figure 13.4: How streaming services change the level

Here what's important though. Since everything is going to be normalized to the same level anyway, there's no reason to crush the mix to make it sound louder as in the old days. In fact, the more dynamic range, the better it will sound – except if your song is meant specifically to be released on radio or CD. These formats will still be played up

against some already loud product so it's best to measure up in those formats.

To look at this issue another way, A-list mixers and mastering engineers don't worry about LUFS levels, so why should you? They create mixes that sounds as good as they can, just as they've always done — though maybe not smashing it as hard. The streaming services will do their thing and release the music at their designated level so, at the end of the day it's not something you have to worry about.

Many music mixers have started using -9 LUFS as a level to shoot for, mostly because it still allows for some reasonable dynamic range while still sounding fairly loud and competitive. As you get closer to 0 LUFS, the dynamic range shrinks considerably, but this can be desirable for electronic or pop music with most songs from those genres coming in at around -6 LUFS.

On the other hand, acoustic music benefits greatly from a higher LUFS level of -12 to -14 LUFS. If you monitor hits from 70s and 80s, you'll find that most tend to fall into this range (or even lower), hence the greater dynamic range. Then again, these songs were made long before mixers automatically placed a compressor across the mix buss as is the norm today.

You still might want to break out that LUFS meter and find a level that you're comfortable mixing at, but at this point, a dynamic range meter is a far better friend.

TIP: More compression and/or limiting will boost your LUFS level. This should be done with caution as higher compression can also introduce audio artifacts that are clearly undesirable.

Checking The Sound Of Streaming Platforms

While you get no advantage for mixing to the playback level of an online music distributor, you might want to hear what your master will sound like when it hits Spotify or Apple Music. Luckily you can now do this without having to post to the platform.

Several developers now offer plugins that will play back your mix at the actual platform level using the identical codec. For instance, you can pre-hear what your song will sound like when posted to Spotify, Amazon Music, Apple Music, Deezer, Tidal, and most of the other major music distribution platforms.

Examples of these plugins include Nugen Audio's Mastercheck Pro, and Adaptr Audio Streamliner (see Figure 13.5).

Figure 13.5: Adaptr Audio Streamliner

Courtesy Adaptr Audio - Plugin Alliance

Alternate Mixes

Before most engineers joined the DAW revolution, alternate mixes were a way of life for mixers to cover themselves in case a change in the mix was deemed necessary at a later time. Renting the studio and then setting up an analog console and all of the outboard gear just to raise

the vocal 1dB in the bridge was both expensive and a time- consuming hassle. Plus, the setup was so inaccurate that the mix might not sound the same the second time around. All that changed with DAWs.

Now that so many mixers choose to mix at least partially in the box, mixes can be instantly recalled and fixes done in a matter of minutes. It's fast, cheap, and easy. As a result, alternate mixes are no longer the norm except for the analog diehards or part of the delivery package specified by a record label. That said, it's important to understand about the most commonly used alternate mixes, because a producer or record label may still ask for them.

"There are just never enough options for some people. When you deal with major labels and managers, there's constantly this thing of, 'Can we get one with this in the second verse?', so I supply many levels of vocal up and vocal down."

—Joe Chiccarelli

Different Types Of Alternate Mixes

As we just discussed, alternate mixes were originally created to avoid having to redo the entire mix again at a later time because of a requested change to a mix. This meant that any element that might later be questioned, such as a lead vocal, solo instrument, background vocals, or any other major part, was mixed with that part recorded slightly louder and again slightly quieter. These variants are referred to as the "up mix" and the "down mix" and the differential increments were always very small; usually not much more than 0.5dB to 1.5dB.

With alternate mixes available, it's also possible to correct an otherwise perfect mix later by editing in a masked word or a chorus with louder background vocals. Many times an instrumental mix is used to splice out objectionable language, especially in the rap and hip-hop worlds.

Although many record companies request more or different versions, here's a version list for a mix that the label for a rock or country artist might typically ask for. Other types of music will have a similar version list that's appropriate to the genre.

Album Version

Album Version with Vocals Up

Contemporary Hits Radio Mix (softer guitars)

Album-Oriented Radio Mix (more guitars and more drums)

Adult Contemporary Mix (minimum guitars, maximum keyboards and orchestration)

TV Mix (minus the lead vocal)

Instrumental-Only Mix

Vocals-Only Mix

The band, producer, or record label may also ask for additional versions, such as a version without delays on the vocals in the chorus, more guitars in the vamp, or a version with the bass up. There is also a good chance that any singles will need a shortened radio edit as well.

Thanks to the virtues of the digital audio workstation and modern console automation, many engineers leave the up and down mixes to the assistants, since most of the hard work is already done.

"I still provide alternate mixes so I don't have to recall anything. Ninety-nine percent of the time recalls are not necessary anyway. The funny thing is that if you give them a vocal up, they'll use it. Give them a vocal even higher, and they'll use that. I still do it, but it's mainly for me for 'just in case.' I do not want to come back to remix. Once I'm done with a song, I've heard it so much that I don't want to hear it ever again."

—Benny Faccone

"I usually only give them the main mix, the TV mix, and the instrumental."

—Jon Gass

"I used to deliver a vocal up a dB and a dB and a half, but it's really not necessary anymore. I now deliver a lead vocal a cappella, a background a cappella, a TV track, and a main pass. Whoever is mastering the mix can put all those parts together to make a fix, as opposed to me delivering a lot more mixes."

—Bob Brockman

Stems

First, let's get the definition of "stems" out of the way. A stems mix is a set of stereo submix files that usually consists of a stereo bed and individual stereo tracks of the most important elements complete with effects. These might include:

Drums

Bass

Vocals

Background Vocals

Guitars

Keyboards

Sometimes the stems might just boil down to the instrumental track with a separate stems track for the lead and background vocals - it all depends on what you're asked for, or what you think might be useful later so you don't have to do the entire mix again.

Many musicians and artists today refer to "stems" when they really mean "tracks," which are all the individual track files that make up a mix. Stems are stereo groups of these tracks complete with all the effects.

This allows for an easy remix later if it's decided that the balance of the lead vocal or the solo is wrong. In extreme cases, some mixers have resorted to the use of stems to keep everyone (mostly the record company) happy.

Stems are widely used in film mixing because a music mixer usually can't tell what may be too loud or masked by the dialog or sound-effects elements of the movie. The stems mix gives the dubbing mixer more control during the final mix, if required. It's typical for the dubbing stage to ask for a stereo (or even surround) rhythm submix, a submix of any lead instruments or voices with effects, the bass by itself, and any instruments with a lot of high frequencies, all isolated on their own submix.

"I still supply a vocal up and a vocal down, an a cappella, and lead a cappella, and instrumental, and then I'll create stems for them so they can re-create any other combination for themselves later if they want."

—Jimmy Douglass

10 Steps For Mixing With Mastering In Mind

One of the things that mastering pros complain about is that so few mixers actually think about how the things they do while mixing might affect mastering. Mastering is the final creative step in the production process where a mix can be altered, but the mix won't necessarily be improved unless you help the mastering engineer out by following some very simple tips when you're mixing.

Regardless of whether you master your final mixes yourself or take them to a mastering engineer, things will go a lot faster if you prepare for mastering ahead of time. Nothing is so exasperating to all involved as not knowing which mix is the correct one or forgetting the file name. Here are some tips to get your tracks "mastering ready".

Don't over-EQ when mixing. A mix is over-EQed when it has big spikes in its frequency response as a result of trying to make one or more instruments sit better in the mix. This can make the mix tear your head off because it's too bright, or have a huge and unnatural sounding bottom. In general, mastering engineers can do a better job for you if your mix is on the dull side rather than too bright. Likewise, it's better to be light on the bottom end than have too much.

Don't over-compress when mixing. Over-compression means you've added so much mix buss compression that the mix is robbed of all it's life. You can tell that a mix has been over-compressed not only by its sound, but by the way its waveform is flat-lined on the DAW timeline (like on the right side of Figure 13.4). You might as well not even master if you've squashed it too much already. In general, it's best to compress and control levels on an individual-track basis and not as much using the stereo buss except to prevent digital overs.

Having the levels match between the songs of the album is not important. Just make your mixes sound great. Matching levels between songs is one of the reasons why you master in the first place.

Getting hot mix levels is not important for mastering. Make it as loud as you need to make it to be competitive or for client approval but leave it to the mastering engineer to get that extra 5% in level gain, assuming that's what you need. Again, it's another reason why you master.

Watch your fades and trims. If you trim the heads and tails of your track too tightly, you might discover that you've trimmed a reverb trail or essential attack or breath. Leave a little room at the beginning and end of the track and perfect it in mastering where you will probably hear things better.

Make sure to export the highest resolution mixes you can. This simply means to export or bounce your mixes at the same resolution as the tracks were recorded at. If the tracks were cut at a sample rate of 96kHz/24 bit, that's the resolution your mix should be. If it was recorded at 44.1kHz/24 bit, that's the resolution your mix should be.

Alternate mixes can be your friend. A vocal up/down or instrument-only mix can be a life-saver when mastering. Things that aren't apparent while mixing sometimes jump right out during mastering and having an alternative mix around can sometimes provide a quick fix and save you from having to remix. Make sure you document the alternate mix files properly so they're not confused with the real master.

Check your phase while mixing. It can be a real shock when you get to the mastering studio and the engineer begins to check for mono compatibility only to hear the lead singer or guitar solo disappear from the mix because something in the track is out-of-phase. This was more of a problem in the days of vinyl and AM radio, but it's still an important consideration since many so-called stereo sources (such as television) are either pseudo-stereo or only stereo some of the time. Check it and fix it before you get to the mastering facility.

Know your song sequence. Song sequencing takes a lot of thought in order to make an album flow so you really don't want to leave it until the last minute, which is to say the mastering session. If you're cutting vinyl, you need two sequences - one for each side. Remember, mastering for CD or vinyl can't be completed without the sequence. Also, cutting vinyl is a one-shot deal with no undo's like on a workstation. It'll cost you real money every time you change your mind.

Have your songs timed out. This is important if you're going to be making a CD or vinyl record. First, you want to make sure that your project can easily fit on a CD, if that's your release format. Most CD's have a total time of just around 74 minutes. When mastering for vinyl, cumulative time is important because the mastering engineer must know the total time per side before he or she starts cutting. Due to the physical limitations of the disc, you're limited to a maximum of about 25

minutes per side if you want the record to be nice and loud with the least amount of noise.

Mastering

Technically speaking, mastering is the intermediate step between taking a mix from the studio and preparing it for replication, but it's really much more. The old definition of mastering is "The process of turning a collection of songs into an album by making them sound like they belong together in tone, volume, and timing (spacing between songs in physical mediums such as CD and vinyl)."

The new definition applies more to the streaming world that we live in. In this case it's "The process of fine-tuning the level, frequency balance, and metadata of a track in preparation for distribution."

Contrary to what many plugin developers would have you believe, mastering is not a plugin or a device that you run some music through and have it come out mastered. It's a highly specialized craft that, when done conscientiously, relies on the engineer's skill, experience with various genres of music, and good taste.

In the early days of vinyl, mastering was a black art practiced by technical curmudgeons who mysteriously transferred music from the electronic medium of tape to the physical medium of vinyl. In fact, for that reason they were originally called "Transfer Engineers." It didn't

take long for mixing to become more sophisticated, and so did mastering.

Mastering engineers soon found ways to make vinyl records louder by applying equalization and compression. Producers and artists began to take notice that certain records would actually sound louder than others on the radio, and if it played louder then it sounded better, and maybe even sold better. With that, a new breed of mastering engineer was born, this one with some creative control and the ability to influence the final sound of a record rather than being just a functionary transferring a song from medium to medium.

Today’s mastering engineer doesn’t practice the black art of disc cutting much, but he or she is no less the wizard as he continues to shape and mold a project.

Why Do I Have to Master, Anyway?

Mastering is considered the final step in the creative process because it’s the last chance to polish and fix a project. It’s interesting to note that almost all of the major record labels and most of the larger indie labels still choose to master all of their projects with a major mastering house (see Figure 13.5), even though extensive mastering resources are widely available to just about any engineer.

Figure 13.5: Oasis Mastering, Burbank, CA.

A project that has been mastered, especially at a top-flight mastering house, simply sounds better. It sounds complete, polished, and finished because the mastering engineer has added judicious amounts of EQ and compression to make the song even bigger, fatter, richer, and louder. He or she has matched the levels of each song in an album so every one has the same apparent level. They've fixed the fades so that they're smooth. They've inserted the spreads (the time between each song) so the music flows seamlessly together on a CD or vinyl record. They've sequenced the songs so they fall in the correct order, and made all the songs blend

together into a cohesive unit. They've proofed your master before it goes to the replicator to make sure it's free of any glitches or noise, and made and stored a backup in case anything should happen to your cherished master. All this happens so quickly and smoothly you hardly know it's happening.

Why can't you just use the "mastering" plugins on your workstation instead of going to a high-priced commercial mastering facility? Besides the points stated above, there are many reasons why a commercial mastering studio produces a better product than home or self-mastering.

First of all, it's better equipped, with super high quality D/A converters, ultra-smooth compressors and equalizer plugins (and even some hardware, too), and an exceptional monitoring system in a precision listening environment. The monitor system alone at one of these facilities may cost about as much as a brand new luxury vehicle. Cost isn't the point; quality is. You can rarely hear what you need to hear in most studios (even the biggest and best) to make the adjustments that you need at this point in the production chain.

Though we often obsess about the mastering hardware, the engineer is the real key to the process. Mastering music is all he or she does day in and day out. They have big ears because they master for at least eight hours every day and know monitors the way you know your favorite instrument. Also, his or her reference point of what constitutes a good sounding, balanced mix is finely honed due to spending hours and hours on the best-sounding (and worst-sounding) mixes of each genre of music.

While all of the above might seem like a way to discourage you from doing your own mastering, that's really not the case. In fact, the reference point of how the pros operate and why they're so successful is important to have. From here, you can determine whether you're better served by doing it yourself or using a pro.

TIP: If you attempt to master your own mix, it's best to use a different set of speakers from what you mixed on, because otherwise you'll only be compounding any frequency-response problems that the speakers might have.

How To Prep For A Session With A Mastering Engineer

Having your mix polished by a mastering engineer is a great experience whether you attend the session or not. Actually attending the session might cost a little more, but you'll get to hear what your mix sounds like in a precision listening environment, and you'll get to see the mastering engineer work his or her magic first-hand.

That said, communication with your mastering engineer is essential to getting the best results and to avoid having your mix rejected. Yes, this happens — even to some of the best mixers in the world — when the mastering engineer sends a note there are some problems that would be better addressed in another mix. Mastering engineers can be wizards but they're also well aware of their own limitations.

The best way to prep for a mastering session is to follow these two recommendations:

Contact your mastering engineer while you're mixing. Don't wait until the mix is finished to make contact. Call the mastering engineer and ask to send a rough mix for evaluation. The mastering engineer will listen, then get back to you with recommendations about how you can improve your mix so it breezes through mastering. You won't be charged for this. In fact, mastering engineers are happy to do it because makes their job easier.

Ask the engineer what kind of master file they prefer to get. The mastering engineer will tell you the level, how much buss compression and limiting, the file format and delivery method he or she needs to best work with your mix.

TIP: Caution: be careful about changing your mix for the mastering engineer if a client has already signed off on it. The resulting master may sound different enough that the client might reject it.

Online Mastering

Like mixing, mastering costs vary a great deal. The costs for mastering with an A list mastering engineer can vary anywhere from about $250 to about $500 an hour. You can expect the overall price for mastering an album at a top facility to come in anywhere from around $1,000 to around $5,000.

However, there are now some very adequate alternatives to spending a lot of dough on mastering. By far the cheapest is to use one of the automated online mastering services like Landr, eMastering or Aria. The prices of these services vary anywhere from as low as $4 a song to a monthly fee of around $100, where you can upload an unlimited number of songs. Depending upon the mix, the results can be surprisingly good, and you're usually allowed to try different mastering settings for just a single price so you're able to choose between something that's more aggressive or more dynamic.

Another alternative is to submit your finished tracks online for mastering at some of the top mastering houses in the world like Abbey Road or Sterling Sound. You won't get one of their senior engineers nor can you attend the session, but the job will be done by a well-trained junior engineer at a price that's considerably lower than the facility's normal rates.

Self-Mastering

We know it's a lot better to have a pro master your recordings than to try to do it yourself, but that's not always possible. Usually, it's a matter of money. In some cases, a professional mastering job may cost more than the project's entire production budget. If you're just starting out you're probably on modest budget so your best option may be to master your music yourself.

The key to doing your own mastering is to be gentle. If you feel the need to add a lot of anything, be it EQ or compression, it's probably best to go back and mix the song again. This happens a lot more than you might think. You'd be surprised at just how little processing a mastering

pro actually adds yet still gets big results. For instance, if you feel that you need 3dB more of anything, it's probably time to rethink what you're doing.

With this in mind, you can do a great job mastering your project if you follow these two points:

Concentrate less on EQ and more on level

Make sure the relative level (the level you hear; not what's on the meters) is the same from song to song of an album

Following these two simple guidelines will make your mastering results better than you'd ever thought possible.

Mastering Your Songs In 6 Steps

When I began writing the latest edition of The Mastering Engineer's Handbook, one of the things I wanted to learn from the current masters of mastering was how they approached a project. In other words, what were the steps they took to make sure a project was mastered properly. Interestingly, the majority of them follow six primary steps, some consciously and some unconsciously.

1. Listen to all the tracks

If you're listening to a collection of tracks such as an album, the first thing to do is listen to brief durations of each song (10 to 20 seconds should be enough) to find out which ones are louder than the others, which ones are mixed better, and which ones have better frequency balances. By doing this you can tell which songs sound similar and which stick out. Inevitably, you'll find that unless you're working on a compilation album where all the songs were done by different production teams, the majority of the songs will have a similar feel to them, and these are the songs to begin with. After you feel pretty good about how these come together, you'll find it easier to get the outliers to sound like the majority rather than the other way around.

2. Listen to the mix as a whole, instead of hearing the individual parts

Don't listen like a mixer, don't listen like a producer, and don't listen like a songwriter. Good mastering engineers have the ability to divorce themselves from the inner workings of the song and hear it as a whole, objectively, just like the listening public does.

3. Find the most important element

For most modern radio-oriented songs, the vocal is the most important element. One of the most important jobs for a mastering engineer is making sure the vocal can be distinguished clearly.

4. Have an idea of where you want to go

Before you start twisting knobs, try to have an idea of what you'd like the track to sound like when you're finished. Ask yourself the following questions:

Is there a frequency that seems to be sticking out?

Are there frequencies that seem to be missing?

Is the track punchy enough?

Is the track loud enough?

Can you hear the lead element distinctly?

5. Raise the level first

Unless you're extremely confident that you hear a wide frequency spectrum on your monitors (especially the low end), concentrate on raising the volume instead EQing. You'll keep yourself out of trouble that way. If you feel that you must EQ, remember to be gentle with it, as you shouldn't need a lot (meaning more than a couple dB's at any one frequency at most) to hear the difference.

6. Adjust song levels so they match

Another critical job in mastering is to take a collection of songs (like an album) and make sure they each have the same relative level. You want to be sure all the songs sound about the same level at their loudest. You accomplish this by listening back and forth to all the songs and making small level adjustments as necessary.

Following the same steps as the mastering greats suggest will ensure that your project will sound better and you'll avoid some of the common pitfalls of mastering your own material.

Remember: Even if you can't get the songs to sound just like your best sounding CD, your mastering job will only be considered "pro" if you can get all the songs to sound consistent in tone and volume!

Self-Mastering Tools

Today, there are a lot of great mastering tools available and they're more powerful than the most sophisticated and successful mastering engineers could have envisioned only 10 years ago.

Some mastering plugins are easy to use but complex under the hood. These include iZotope Ozone 9, the IK Multimedia Lurssen Mastering

Console (see Figure 13.4), Waves Abbey Road TG Mastering Chain, or the bx-masterdesk from Plugin Alliance. Other plugins have a space-shuttle feel to them like the Howie Weinberg Mastering Console from Acoustica which, though very powerful, can be a bit too much for a beginner.

Figure 13.6: Lurssen Mastering Console

Courtesy IK Multimedia

Some plugins are just mastering style compressors like the Shadow Hills Mastering Compressor or PSP Mastercomp, while some EQ plugins, like the like the Hofa IQ-Series V3, are commonly used by mastering engineers at the highest level.

Many of these tools are meant to be used as plugins on the master buss or inside the DAW session, while others can be used as stand-alone apps that sit on your desktop. This interface separation can actually be valuable as it gets your head out of the mix and puts it squarely in a mastering space.

Regardless of which ones you use, remember that a little goes a long way. If you feel the track needs a lot of mastering adjustments, you might first want to re-examine your mix.

PART II

THE INTERVIEWS

Bob Brockman

"Bassy" Bob Brockman has a wide range of awards and credits, including more than 30 Grammy nominations with two wins, and an Oscar nomination. His many credits include Mary J. Blige, Toni Braxton, the Notorious B.I.G., Babyface, Aretha Franklin, Al Green, The O'Jays, Brian McKnight, Jodeci, Faith Hill, Korn, Laurie Anderson, Vanessa Williams, Christina Aguilera, Diddy, Herbie Hancock, the Fugees, Santana, and Sting. He's very much of the "old school/new school" in that his formative years as an engineer were spent in the analog world, but he's quite at home in the current digital one as well.

Can you hear the final product in your head when you begin to mix?

Yeah, probably. I think that I probably make some subconscious and non-verbal judgments when I first hear a song. I make a judgment on style and then go through a couple hours familiarizing myself with all the parts, then I try to see what's really crucial and what could be wallpaper. I then find whether there's something that's really important that I should make the listener aware of.

The first 20 years of my career I had a producer standing right next to me, telling me what parts were important. It's less so now because I see fewer people. I get sent digital files, and I sort of end up making those mix/production decisions on my own and end up delivering a more or less finished mix to the producer or the band; then I'll get notes on what to tweak.

Are you mixing on a console or in the box?

I can mix in the box if I have to, but it's certainly not my preferred way of mixing. What I'm into is a sort of hybrid mixing. I have a Neve 8816 [analog console] with 16 channels coming out of an Avid 192 D/A with an Alan Smart C2 compressor across the mix buss. Sixteen channels of analog makes a big difference to me in terms of power and depth of field. I still do mix quite a few things in the box, though, especially when I'm out traveling.

How much of the DAW do you use?

I'm very deep into the whole digital mixing process and do all of my work in Pro Tools. The plugins have stepped up a lot in the last few years, with the distortion and saturation plugs having improved immeasurably. They're now more transparent and not adding a lot of phase shift or distortion when you insert them. That was my problem with plugs before and why I would tend not to use them on phase-dependent things like drums and guitars.

At a certain point, maybe the seventh or eighth hour of the mix, the whole thing would start to sound crunchy to me, so I would go in and bypass the plugs and realize that I was using them as a crutch to make things speak. Once you get things dialed in, by the end of the mix you don't need them as much, so there's a lot more sonic purity.

I often encourage young mixers to bypass their plugs and listen to what they have, especially in a program like Logic, where when you open up a session it's already got three or four things inserted across every channel as a default.

There are certain things that you can’t avoid, like de-essing. If you want to remove the sibilance from a vocal, you’ve got to use a de-esser, and the Massey de-esser is probably my favorite of all of them on the market. In fact, I’m a really big fan of Steven Massey’s plugs. I use his L2007 [mastering limiter] a lot.

What are your go-to plugins?

I really like the Brainworx stuff, and I use a lot of the Waves stuff, like the API and the Jack Joseph Puig and Chris Lord-Alge Signature stuff. They don’t always sound exactly like their analog counterparts, but that’s not necessarily important. They definitely lend a power and tone change to what you put across them. I tend to use the CLA LA-3 plug more across the toms, snare, kick, guitars, and sometimes bass, because it has an interesting sound to it.

For reverbs I use the [Audio Ease] Altiverb a lot, as well as the stock Pro Tools Air Reverb, which I’ve made a lot of presets for over the years, so I have one for every situation. I also use the Lexicon vintage reverbs a lot.

Do you have certain effects that you always set up before you begin a mix?

I typically transfer all of my effects from one song to the next. I’ll usually use an [SoundToys] EchoBoy or a [Massey] TD5 for delay. The [Waves] H-Delay and the [PSP] Lexicon PCM 42 are really nice as well. I usually have four or five delays, which vary from very tight to slap delays to timed things. I tweak the timing so it’s either pushing or dragging a bit behind the beat. I usually have four or five reverbs all plugged in as well. I don’t have any analog effects processing. It’s all done in the box.

I do have a pair of Neve 1073s that I might insert across the stereo buss, but for the most part I'll just leave the equalization to the mastering guy. I try to get the EQ and the sound from what I'm doing to the individual tracks in the mix. I've never been much of a user of equalization over the years. I've worked with a lot of master buss equalizers like the Massenburg stuff, but there are so many equalization things that happen to the sound just by making adjustments on the Alan Smart [SSL-clone compressor]. It's such an amazing compressor with the way it grabs the low end and accentuates certain parts of the midrange or upper midrange, depending upon how fast or slow and the ratio.

I usually spend the last two hours of my mix not doing much mixing but listening and then making little adjustments to the master buss compressor and hearing what the impact is to all the parts. I definitely don't have a stock compression setting. I'm always moving the setting around on everything that I do. Each song has to have its own contour, I guess.

How hard are you hitting it?

That depends on the music. If I'm doing a dance record, I'm probably hitting it pretty hard. If I'm doing an aggressive rock record, then I'm sinking into it about 3 or 4 dB. If I'm doing something much more open or acoustic, I'm barely hitting it. Most of the effect is how it's putting the low-frequency information in check, which it does without the meter moving at all. Even when you're hitting it very lightly, it still has a dramatic effect on the music.

One thing I really like is The Glue [from Cytomic Sound Music Software], and I use it a lot. It’s one plug that reminds me of the Alan Smart a little bit because it has a very dramatic effect. I also use the Steven Slate Master FG- X [Virtual Mastering Processor] plug, which is amazing. There was a time in the not-too-distant past when you would insert a plug across a channel and go “eh,” because as you would open it up it sounded worse and worse. That has changed a lot in the last two or three years in plugin development.

Where do you start your mix from?

I’ve always mixed the whole thing together at the same time. In the old days I would set up everything across the console and very quickly sort of push things into place manually. I do the same thing now in Pro Tools, although I don’t think it will ever be as intuitive a process in mixing in a DAW as on a console. There’s something about sitting at a console with a bunch of faders and being able to grab something and move it, which is hard to replicate with a mouse, but I’ve gotten used to that workflow these days.

Another thing is I’ve been mixing with my eyes closed for most of my career. It was something that I started doing when I was 23 or 24 years old. I think I have an easier time visualizing the three-dimensional panorama that’s coming out of the speakers. It’s also a tool to help me to localize things. Somehow when I close my eyes it’s easier for me to see an instrument or vocal by removing my eyes from the equation altogether.

Do you use a work surface as well?

I was using a couple of the Euphonix MC Mix controllers, but it got to the point where they were taking up a lot of space on the desk. I do a lot

of work in the DAW graphically, especially in Pro Tools X, where I've gotten into Clip Gain. Automating vocals may have a thousand small moves to make sure the apparent volume of every single word is the same. The Clip Gain has a more natural sound to it, I think because you're adjusting the level of the vocal before it goes into the signal chain, whereas with the volume-based automation, you're adjusting it after it's already gone through the plugins. When you raise it up through Clip Gain, I think it behaves more like a console.

What monitors do you use?

I've been mixing on KRK E3s for 13 or 14 years. I use them with a 12-inch subwoofer that I set very lightly. With a lot of modern music there's a lot of sub information that's going on, and it helps if you can hear it a little more clearly.

How loud do you listen?

I listen pretty loud [laughs]. It's loud enough that I drive everybody out of the room for the first couple hours of the mix. I think it's important to know what the music sounds like loud, but once I've made the global decisions about the low end, then I'll usually go down pretty quiet, and that's where I tend to stay for the remainder of the mix. I might bring it back up when I'm getting closer to print just to make sure that there isn't anything that's too harsh on any of the really loud things of the mix.

What do you deliver to the client these days?

I used to deliver a vocal up a dB and a dB and a half, but it's really not necessary anymore. I now deliver a lead vocal a cappella, a background a cappella, a TV track, and a main pass. Whoever is mastering the mix

can put all those parts together to make a fix, as opposed to me delivering a lot more mixes. I always thought it was absurd to deliver a mix with the lead vocal up a dB and half because it's usually not what you need. You usually only need a breath or a word or a chorus where it's just not loud enough. I don't think they ever got used anyway. I did that for about 15 years, but not anymore.

Has your philosophy changed from when you started to now, especially now with the digital workflow?

Yeah, it's changed a lot. There's so much detail in every mix now. I find myself automating almost every aspect of everything in the mix, whether it's changing the EQ of the backgrounds as they go into the chorus so they punch through, or automating the sends and returns to each of the effects. I'm also constantly filtering things like delay returns to make them brighter or duller at different sections of the song. It's now so much more involved and detailed than anything that I would've ever been able to accomplish on a normal analog console.

When I go back and listen to mixes that I did 15 years ago, I think, "Wow, that record is kind of fat," because it was done on an analog desk, but it's certainly not as involved and detailed as the mixes are now. It was enough to have a mix that had a great sound and a great feel to it 15 years ago, but everyone expects a lot more precision today. I also find that I'm doing 15 to 20 revisions now, and that just never happened earlier in my career. People just have the ability to endlessly make changes, and because they can, they do.

How long does it take you to do a mix these days?

It's an interesting question because it also begs the question, "What is a mix?" There was always a very clear delineation in the production

world that I grew up in. When you'd do a mix, you wouldn't really do that much to it except maybe sample a snare drum to enhance it or something like that. The record was what it was.

Today there's a lot of cleanup stuff that's sort of expected as part of the mixer's job, which should be a production thing. Production has gotten a lot lazier in the past 5 or 10 years. Now it's unbelievable how much garbage, extraneous stuff, clicks and pops, and unlabeled tracks that you get even in major projects. I don't know whether it's because you can have endless takes so you figure, "I'll just leave it to the mixer because he knows how to do that." Of course we do know how to do that, but it means the mix will take a lot more time.

If I'm mixing one of my productions where I've done the cleanup and organizational stuff ahead of time, I can usually mix it in six to eight hours. I find that with a lot of records that get delivered to me to mix, there are all these problems, like the drums aren't in sync or things have been thrown out of phase, and no one seems to have noticed. Maybe it's because there are too many tracks or maybe they just don't hear it. Other than the mastering engineer, I'm the last guy to see the record, so if those things are going to be dealt with, I have to deal with them, but it gets to be enormously time consuming.

How much do you replace sounds?

I guess it depends on the record and the style. If I have to replace something, I'll use the Slate Drum Tool, which is amazing. It has an incredible library, and it's really the only drum trigger that I've found that does it in time. If I get a rock record where the toms are less than inspiring, I'll replace them. If I get a snare that sounds like it was recorded in a closet, I'll add one that has a nice room sound to it.

More often than not these days, rock records are recorded in some austere conditions, like in a closet in their house. It's not a killer big room drum sound, and that's what you want because listeners have an expectation of how it should sound. You have to do whatever you have to do to get it to a place where people expect it be. It can be a really time-consuming process.

Bob Bullock

Since he moved to Nashville in 1984, Bob Bullock has been one of the town's top engineers, trusted by the likes of Kenny Chesney, Shania Twain, George Strait, Reba McEntire, Hank Williams Jr., and Jimmy Buffett, among many others. Prior to that he saw a different side of the music world working in Los Angeles with acts such as the Tubes, Art Garfunkel, Seals and Crofts, Chick Corea, and REO Speedwagon. Now an instructor at several recording programs in Nashville, Bob continues to be a much in demand mixer. You can read more about Bob and see his complete discography on his website at bobbullock.net.

Can you hear the final mix in your head before you start?

Yes, but I don't necessarily know how I'm going to get there when I begin. I have to listen to the song and all its pieces before I have a vision for it, which is important because if I didn't have a vision of how I wanted it to sound, I'd just keep going around in circles.

Where do you start the mix from?

I start with the drums and bass and get the basic foundation, then add the guitars and piano or anything that would be considered part of the rhythm section. After that feels good, then I put a vocal in, because the style of music that I do is all vocal-driven, so the sooner I get it in the mix, the better. After that, I place any of the ear candy around the vocal and rhythm section.

I build the mix with balance, panning, EQ, and compression; then when I'm happy with all that, I'll start adding effects to give it depth. The final thing I do is add some rides on things like the vocal to build in a broader dynamic.

Do you start with the kick?

I found that it works best for me to keep the entire drum kit on while I'm adding EQ or compression. That doesn't mean that I won't solo the kick or snare, but what I don't do is just solo the kick and work on that for a while, then solo the snare and work on that. It usually starts with the whole kit so I can hear what the leakage is doing and hear what I need to do to make the kick or the snare pop through a little more. After I'm comfortable with the balance, then I'll solo the kick and maybe contour it a bit so it pops a little better, but I still try to make the drums sound like just one instrument.

How much do you do in the box these days?

I'm doing quite a lot in the box. In fact, I've got a couple of racks in the studio filled with outboard gear, but I'm in the process of letting a lot of it go. My new setup is mostly plugins based.

What are your go-to plugins?

I'm really a fan of the UA stuff. They've helped make an old-school analog guy like me embrace digital. I definitely use their EMT 140 and 250 reverbs a lot, the LA-3As because they've always been my favorite guitar compressors, and occasionally the A800 tape emulator. I've got the Slate Digital mix buss plugins [the Virtual Console Collection] that I like a lot because it's not a compressor or an EQ, but it's bridging the gap between them. I don't use it every time, but when I feel that the

mix may be lacking something, the Neve or Trident setting really helps. I like the Waves R-Vox and R-Bass, and I use a lot of the PSP stuff. One of my favorite delays is their PCM 42, and I like to use their EQs and compressors as well. All that said, if I can I try not to limit myself to just the go-to pieces and plugins that I usually use and try to change things up a bit.

What do you use for a controller?

Right now I'm using the Euphonix MC Control. In my old studio I was using some Mackie controllers that had 24 faders, but they were pretty slow because it was MIDI, and the MC controller is faster. I'll probably end up with a larger control surface so I can get more faders, though.

When you were working with Mutt [Lange] on Shania's albums, was it different from anything else you've done?

If I had never worked with Roger Nichols and Steely Dan as an assistant, I would probably say yes, because the biggest difference working with Mutt from most other sessions was the extreme attention to detail. Going back to when I used to work with Roger and Gary Katz and Walter Becker and Donald Fagen [of Steely Dan], the number of hours that used to go into everything was amazing, so to me working with Mutt wasn't all that unusual. For someone who's used to things moving faster, it would probably be torture. Mutt, for all his success, is basically a writer who produces what he writes or co-writes. His approach has always been not that different from Donald and Walter, or Paul Simon or Gino Vanelli or some people like that who put a lot of labor into their music.

We rarely get a chance to craft a record at those levels anymore because the budgets don't allow it. I'm just as much a fan of things that

are done swiftly, but I admire those guys for the patience that it takes to get things perfect.

What's your approach to compression?

It depends on what I'm working on, but I tend to start with more compression and then pull back a good bit of it by the time I sign off on a mix. I always have a least some buss compression on all my mixes because it's expected of us now. When a client hears a mix, he wants it to pop. In the '70s we used to feel that we could control it better in mastering, and even though I add buss compression today, I make sure to leave a little for the mastering engineer to work with.

I always mix toward the buss compression knowing that it will be part of the dynamics, but on the individual tracks, it all depends on what I need to do to have it hit the buss compressor gently. I try to make them controlled and contoured enough so that when it hits the buss compressor there won't be anything extreme going on.

Are you aiming more for dynamic control or the coloring of sound?

For me it's for dynamic control. Back in the days of analog, there was something about mixing through something like a Neve 8068 to 1/4- or 1/2-inch tape that automatically gave it a fatness and dynamic control. In digital, because everything is so clean and clear, it means I have to use a little bit more individual compression to do the same thing. What I'm always trying to do is find ways to get a little of that analog fatness without using analog, and I have to say that using some of these mix buss plugins does that sometimes.

Do you use a lot of effects?

Yeah, I do, but it varies by the project and what the client expects. My personal taste is to use more layers, like using several reverbs to create one reverb sound, or using several short and long delays. My reverbs and effects usually end up coming from four to eight different sources. They'll be short, long, bright, dull, and everything you need to make an environment.

I use effects and reverb like we used to in that I'll set up an EMT 140 plate and have just about everything in the mix go to it a little bit. I guess if I was mixing a Nicki Minaj track I probably wouldn't do anything like that, but because everything I do is more organic, I find that feels the most natural.

Do you have an effects template?

No, I put them in as I go along, but what is common is that I might have a [Universal Audio EMT] 140 set to short and then a few others that might be longer. Same thing with delays. If I want to keep it hidden, I'll just use a shorter delay. If I want more depth, I'll have something short and long and blend them together. I learned a lot of that from [TEC Hall of Fame engineer] Roy Halee back in my assistant days because he used a lot of tape delays to give the mix depth.

Do you ever replace drums?

I'm a real fan of Drumagog to add samples to existing drums, but I rarely replace them. If someone sends me something where the drums are just unusable, then I'll have to, but generally I'll add a sample to add something or trigger an effect.

Do you monitor loudly or have any listening tricks?

When I'm starting off, I build my basic mix moderately loud. When I'm satisfied that everything is where I want it, then I listen pretty softly to see if something pokes out too much.

I'm pretty much a one system, near-field kind of guy now, since I began using the Carl Tatz Design Phantom Focus System (PFS) 4 based around a set of Dynaudio M1 Professional Monitors.

I also use a set of self-powered NHT M00s to listen as well. I either listen to them standing in front of them or even listen from another room. The PFS is definitely the main system, but occasionally I'll listen to the Moos to give me a closer-to-small system comparison to check balances.

As far as listening tricks go, when I have all the songs of an album together, I'll make a CD and drive around in my car listening to it, keeping mental notes of things that I want to change. I personally have tried to get away from listening to things in too many places because it can get a little confusing.

Since so many people are listening on their phones and laptops these days, it helps to have a little bit of a perspective of what that sounds like too, but if we can make something really pristine and hi-fi on the big system, for the most part it's going to translate to those systems as well.

How long does it take you to do a mix?

That can vary, but I got used to having one day per song, although the PFS monitors has sped that up some. In the days of analog I'd start a mix, and later in the day the client would come in and we'd go in circles a little bit and make changes. If we still weren't sure, we'd come back the next morning when we were fresh, tweak it again, then we'd set up for another one. Under those circumstances, my ideal mix schedule would be 12 days to mix if there were 10 songs to do.

I don't mix that way much anymore because I do so much in the box. Now I kind of dabble with all the songs I have to mix for a couple of days, and then I'll fine-tune them one by one. If I like what my real LA-2A is doing rather than one of the pluginss, I'll just print it so I don't have any recall to worry about. Same with any outboard reverbs I might use; I'll just print them to audio tracks. When I mix this way it takes about five to ten hours per song.

Do you have a philosophy about mixing?

The general philosophy that I'd like to pass down is that all these plugins are just tools, and what might work for one song may not work for the next. There's no one preset that's always going to work on everything. Even if there was one, you still have to be wise enough to know when to run with it.

When I worked with Al Schmitt [as an assistant]—and there's no question about the quality of his recordings—he used very little processing. He did it all with balance and the right reverb. With Roy Halee, I would spend all day setting up tape machines for tape delay,

and he also made great records. Humberto Gatica, who I also used to assist, used a lot of outboard gear, and he also made incredible sounding records. What I learned was that it's okay if your method is different from someone else's because it doesn't matter how you get there. Take all this information in, but in the end, use your ears.

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Joe Chiccarelli

Even though he may not have quite as high a profile as many other big-time mixers, engineer/producer Joe Chiccarelli's list of projects is equally as notable as the best of the best. With credits like the White Stripes, Alanis Morissette, the Strokes, Jason Mraz, Tori Amos, Beck, U2, Elton John, Oingo Boingo, the Shins, Frank Zappa, the Killers, Brian Setzer, and many more, chances are you've heard Joe's work more times than you know. In this updated interview, the nine-time Grammy winner describes how analog and digital combine perfectly during his mixes.

How much are you doing in the box these days?

A small portion. I definitely mix in the box when the budget only allows that, but I still love breaking it out on an analog console because it sounds so much better with the stuff that I do. I'm doing rock bands and organic music, and that stuff needs the size and aggression you get from a console. I don't feel that I can get the same results in the box, although I do a lot of premixing there. If you have 20 tracks of background vocals, it's better to premix them to stems there first. Even if I'm mixing in the box, I always try to use some sort of summing network because that improves the sound a lot.

How has the way you work changed now that so much goes on inside the box?

The digital world makes life easier and more cumbersome at the same time. The possibilities are endless, the ability to share sessions for collaboration is phenomenal, plugin technology has really come a long way, and the software has vastly improved. Everyone appreciates the

sonic character of analog, but they don’t want to or can’t take the time for it.

I’ll still mix to analog tape when the budget allows since I love the sound, and I do that on probably 75 percent of the projects I work on because of the glue it adds. Unfortunately, people can’t always afford the cost of tape in the budget, and tape is not as consistent as it used to be.

How long does it take you to mix a track?

It really depends on the material, the number of tracks, and the arrangement. I try to work fast because I find that the longer it takes, the more I get into a sort of myopic mindset and get bogged down with the little details. You miss the vibe and the big picture and just suck the soul out of it, so I like to put it to bed in eight hours or so. In three hours I want it to sound like a record with the basic sounds and feel. In six hours I should have all the balances and it should start to sound finished. After that, the artist will come in for a listen.

As a tool, Pro Tools speeds up the process for you as a mixer, but the state of the world is indecision. I get lots of songs to mix that are 100 or 120 tracks with guitars upon guitars, and they want me to sort it all out. I don’t need someone to print three mics for every guitar that they record. I think that one should have a concept of what the record should sound like and just go for it.

That’s the thing that does take a bit more time these days in that there are so many possibilities, and people don’t make decisions now because they know they don’t have to commit to something until the very end.

In fact, many of them don't even commit at the end of the mix and wind up taking a Pro Tools system into mastering to make the final decision there.

Where do you start your mix from?

I have no system. I really work differently for every project and every different type of music. It's a matter of finding out what the center of the song is or what makes the song tick. Sometimes you build it around the rhythm section; sometimes you build it around the vocal.

Usually what I do is put up all the faders first and get a pretty flat balance and try to hear it like a song. Then I make determinations from there whether to touch up what I have or rip it down and start again from the bottom.

If you're mixing an album, do you vary the sound from song to song or keep it all in the same sonic ballpark?

The approach varies from song to song, but I try to keep the same kind of reverbs and treatment for the drums. I try to keep some level of consistency, but again, I'm also treating every song differently as well. I personally like records that take you to 10 or 12 different places.

Do you add effects as you mix?

I usually try to start out with a flat track and then find the tracks that are boring and add some personality to them.

Do you have a standard effects setup?

Once again, it depends on the music. On some projects I'm not using any reverbs at all, while on some projects I might be putting all my reverbs through SansAmps or some other kind of cheap stuff. Sometimes I may only use a reverb and a delay for a whole record, but there are projects where I might use a plate, a live chamber, and a couple of old-school digital reverbs and tons of delays and plugin delays, so it's really song driven.

Do you have an approach to EQ?

It's weird. I just use whatever it takes for the particular project. It depends on what's been recorded, how well it was recorded, and how much work it needs. Bob Clearmountain is the genius for knowing what to touch and what not to touch, and I think that's really the secret—what to fix and what to leave alone. I find that the more I mix, the less I actually EQ, but I'm not afraid to bring up a Pultec and whack it up to +10 if something needs it.

One thing that I use is a spectrum analyzer that I put across my stereo buss that lets me know when the bottom end is right or the S's are too sibilant. I know what a lot of records look like on the analyzer, so I can tell when the overall frequency balance is right or when it might have some obvious little hole in it. It definitely helps me sort out the 30 to 50Hz portion of the low end before it gets messy.

What's your approach to panning?

What I do is once I have my sounds and everything is sitting pretty well, I'll move the pans around a tiny bit. If I have something panned at 3 o'clock, I'll inch it a tiny sliver from where I had it just because

I’ve found it can make things clearer that way. When you start moving panning around, it’s almost like EQing something. I find that if I nudge it, it might get out of the way of something or even glue it together.

How much are you compressing things?

Compression is like this drug that you can’t get enough of. [Laughs] You squish things, and it feels great and it sounds exciting, but the next day you come back and say, “Oh God, it’s too much.”

I do a lot of parallel compression, where I’ll buss my drums to another stereo compressor and blend that in just under the uncompressed signal. Sometimes what I’ll do if everything sounds good, but the bass and kick drum aren’t locked together or big enough to glue the record together, is take the kick and bass, buss them together to a separate compressor, squish that a fair amount, and then blend it back in. I’ll add a little bottom end to that if the record still isn’t big enough. This helps fit the bass and kick lower on the record and gets it out of the way of the vocal.

On the mix buss I use a number of different compressors; sometimes it’s an Alan Smart, sometimes it’s a Focusrite Red 3, and sometimes it’s something else that seems to work for the song. If I want something that’s soft and warm, I’ll use a tube compressor. If I want something that will give me a little more pop and glue, I’ll use the Smart. In all cases, it’s never more than a dB or two, because it’s just for gluing the mix together. I’m not pumping it up dramatically to make it radio-sounding or anything like that, because I don’t make those kinds of processed pop records. Most of what I do has real singers and songwriters, and they’re not layered to death, so I don’t have to compete with the latest flavor-of-the-month sound or artist.

Do you use more delays than reverbs?

It depends on the project. If it's a slick pop thing, then I might use a lot of reverbs; but if it's a rock band, then I might only use one reverb and maybe half a dozen delays. I'll try to do different things, like put only one instrument in the reverb or put a reverb in mono and pan it in the same space as the instrument. I like the mono reverb thing because it doesn't wash out the track, especially if you EQ the return of the reverb so that it doesn't conflict frequency-wise with the instrument. I've done some fun stuff, such as compress the returns of the reverb so that they pump and breathe with the signal that's there. It gives the reverb a cool envelope that comes up after the dry signal and doesn't fight too much with it.

What are you using for monitors these days?

I've been using the Tannoy AMS 10A's for quite a while, and I usually use those in conjunction with NS10s and Auratones. Every once in a while I'll go up on the big speakers if those are good. I might get my sounds at a pretty moderate to loud volume, but when I'm getting balances it's always really soft. I listen in mono an awful lot and find that's great for balances. You can easily tell whether an instrument or vocal is fighting something else.

Do you have any listening tricks that you like to use?

Yeah, I might walk out of the control room and listen to it right outside the door. Things tend to pop out that aren't as obvious in the control room.

Do you have any go-to plugins?

I would say that the only thing I rely on is the UA stuff, just because I think their emulations are the best out there, and their library has so many options that I can go through super fast to find the perfect sound. I use a lot of their reverb and delay plugins. I like to use the EMT 250 on the snare, maybe the EMT 140 or a Roland Space Echo on a vocal. The dbx 160 has always been one of my favorites on bass guitar, so that's nice to have as a plugin, and the Trident A-Range EQ has always been one of my favorites, especially on electric guitars. Another favorite is the Little Labs Voice of God Bass Resonance plugin, which I love for putting some subharmonic frequency on an instrument.

When you're mixing on an analog console, where do you do most of your automation?

Both on the console and in the box. I find that I do basic moves for dynamics on the console, but all my fine-tuning is done on Pro Tools, especially the vocals. Stuff that requires tiny little maneuvers, I'll do almost all of in the computer.

Having plugins automated is a godsend and so easy compared to what you have to go through on an analog console. I'll automate things like feedback time, release time, or pre-delay time of the plate from section to section, which would be really difficult to do in the analog world.

When you get something into mix, how much cleanup do you have to do?

Too much, usually. [Laughs] One issue is organization, because the track labeling is often really poor, and I find I'm spending hours of prep time before I can even get into a mix. I get lots of tracks that are just labeled "Audio 1," "Audio 2," or "Bob 1" or "Bob 2." I don't care that it's Bob singing. I need to know exactly if it's a high harmony in

the chorus. Same with the guitars. I'll get some that will be labeled "Matt" and "Jim." Now I have to play them to find out that they're even guitars, first of all, and then figure out what role they play in the song. I find that I have to spend two or three hours before I can sort it all out, even before I begin to mix.

Do you do much replacing drums?

I would say I do augmentation rather than replacement. If I use samples at all, I'm only mixing them 20 or 30 percent of the original, just to give them more consistency and more help to cut through the mix better. If I do it's with the Slate Trigger, which I've found to be the most consistent of all those kinds of programs, although I find I still have to tweak it a bit.

Are you doing many alternate mixes?

Constantly. There are just never enough options for some people. When you deal with major labels and managers, there's constantly this thing of, "Can we get one with this in the second verse?", so I supply many levels of vocal up and vocal down. It's just so much indecision. Back in the day when the technology was limited, you knew what you wanted and you just went for it, and that was it. Jack White, bless his heart, sticks with eight tracks. That's the record. Done.

17

Richard Chycki

Richard Chycki has worked with everyone from Rush to Dream Theater to Aerosmith to Mick Jagger and more. In fact, he has 26 gold and platinum records from Rush alone! While he's worked on projects for many mega-stars, the last Dream Theater album may have shattered a record for complexity. While a home project that hits 50 or 60 tracks may feel immense, it's also interesting to compare that to what a really large project can be these days.

How much mixing in the box do you do?

Quite a lot. If I'm doing a surround project, then I do that entirely in the box. If it's in stereo, then I bounce between the box and and an SSL Sigma that I use for a hybrid approach.

What's your approach to mixing in surround?

I discuss it with the client first thing in a project because you get a better result if you record some extra elements with that in mind. Any additional microphones can give you some additional space to make it more authentic sounding so you feel like your in the room with the band. I don't use fold downs for the stereo mix, I'll do stereo separately so that the client is happy with the imaging, then I'll put it out to surround with all the extra mics. As a result, I end up mixing the album twice, more or less.

For surround for a live album I like to keep a very strong audience perspective, so the band is in front and there are reflections in the rear.

I'll spend a lot of time just listening to the hall mics to see what's going on and what's bouncing into the rear mics. I'll try to get a good recreation of what the house sound mixer is doing, along with the audience, so it really feels like you're there.

From a studio point a view it's a different approach, especially with a band like Rush where there's a chance to experiment. They want to maintain the feel of the original mixes, so there's definitely a heritage factor there, but try to get it interesting as well. When I mixed "Limelight" everyone thought that Geddy came into the studio to add a vocal, but it was just a harmonizer tuned down a fifth that was on the original but way down in the mix. When I heard it in the multitrack I thought I'd put it right in the rear speakers so every would hear it, and they did [laughs]. They thought we were re-recording it and it was sacrilegious somehow. It was always there on the album, and if you pop on a good set of headphones you can hear it, but when you remove it from the density of the front of the mix, it's a new experience.

How do you approach the LFE?

For live I'l put a little bit of kick drum, a little bit of bass, maybe a touch of guitars for the palm mute, and then I'll use it especially for pyro. I'll get in recordings where the visual of it is so amazing with a bomb going off and all you'll hear is a pip. I'll push a bunch of that down there so it sounds like what if looks like.

Are you adding sound effects?

That's been recording dependent. There are times when I didn't have to, and there have been times where, for whatever reason there's no low end, I'll add something to give it some power, but it's only in the LFE so your eyes and ears match up.

How do you handle the audience because there's two schools of thought. One is that you add in at the beginning and it becomes part of the bed of the song, and the other is that you add it in at the end as an enhancement.

That changes on a case by case basis. I'm looking more at the room ambience than anything. If it's something that I'm recording I'll go out and put 10 room mics up. What we're looking for there is a really great ambient sound, so we don't much worry about the audience. That's the biggest thing for me to achieve. When I turn this up, is it a good ambience and does it support the direct signal from the stage? If I have a good room sound, then the audience is usually on my side. If it's a crappy sounding room then it's not going to be a good mix when it's done.

Do you have certain plugins that you always use?

I love all the of the McDSP plugins. I use a lot of the Sonnox plugins, a lot of Universal Audio, and a few Waves as well. The UA SSL plugins sound really great as do the plugins from SSL. I really enjoy that SSL console grab and bite, and what I try to do is recreate than in the box.

What's your approach when you're mixing something that you didn't track? Do you listen to the rough mixes or just go in cold?

I find that speaking with the artist is more beneficial than listening to the rough mix. So I try to find out what they want because if there's a mix already in existence then they're not happy with it, and I don't want to listen to something that they're not happy with.

If they mention something about the rough in our preliminary conversation, then I'll refer to the mix to find out what it is, but I find

that if I learn what the band is about and where they want to go with it, it's easier to go from there.

How long does it take you to do a mix?

If everything is clean, sometimes songs mix themselves. I've had some that were 4 hours and others that were 3 days. There are no rules.

The great thing about mixing in the box is you can do what you do and the vocalist will come in and ask for the chorus to come up a dB. you just open it up and 90 seconds later, you're printed, so there's not that physical recall that can be incredibly time consuming.

Now the other side of that is that you've got all the minutiae that comes with it, like "The drummer says that the 37th high hat hit has to come up." [laughs]. You can have a situation where instead of having one, maybe two recalls, all of the sudden you're into 12. The time that the mix is signed off on is getting longer because of the ability to do last minute changes.

What do you do when you have a lot of tracks?

I end up doing some mixing down before we mix. I may take as long as a week to go through and reduce the tracks and bounce it down to stereo if the elements are similar. If there's 3 different flavors of piano in a song, I'll bounce it down to a stereo pair, that sort of thing. You tame it one way or another. It's no different than a movie. You comb through them a few times and reduce it down until you get what you hear in the theater. Slowly but surely it will happen. On the last Dream

Theater album (The Astonishing) the biggest song track-wise topped out at 577 tracks!

That's mind-boggling.

It's mind-boggling for musicians. It's funny because I talk to anybody involved in making records and they're impressed. Anybody that's involved with making movies just goes, "Ha, rookie. That's small." All the movie guys are going, "Hey, I guess you're ready to come over to the dark side and work on films, brother."

I should clarify that when I say its 577 tracks, I don't mean it's 577 different elements. There's quite a few mics involved to record an orchestra or choir, and David's [arranger David Campbell] been doing this a long time, so every recording had extra microphones set up for contingencies. For instance, there's extra spot miking in case you have to pull something up in the mix of the orchestra.

During the mixing phase I was using three Pro Tools rigs, all locked together with word clock and time code. We'd do submixes on all of the three rigs, then those submixes were feeding an SSL Duality console at Germano Studios in New York City.

Was there any problems syncing three machines up?

No, they actually played very nicely together once all of the initial work clock issues were out of the way. Everything must use high quality cables. If you use any bad cables it's woefully apparent with word clock. Most of that work was just getting word clock set up and then getting time code to everything so all of the machines knew where they

were at. The lock up was so fast because it's not mechanical like the old days on tape.

Were most of your effects in the box or were they external?

For this project I actually brought a lot of the effects from my studio. Things like a Lexicon PCM96X and 300L, a Bricasti M7 are the core of the spacial effects. Not to say that we didn't use things in the box. We actually did that a lot on the last record, but we were trying to get a different texture this time.

Can you hear the difference?

The funny thing is, yes, I did hear the difference. They sit in the mix really well. In a sparse mix it becomes less of an issue, but if the mix becomes really complex, you do hear it. You can have more elements coexist in the same soundfield rather than have them jumbled together. I wish I knew what it was because I would bottle it and sell it to everybody [laughs]. There is something there. Is it psychoacoustic? I don't know, but it's one of those things that felt right, so we went with it instead of trying to fight it and say, "We have to do this in the box."

Did you find that the mix went faster or slower because you were on a console rather than in the box?

Being completely in the box is great because you can stem out more at a time. I would leave my assistant to do multiple passes of stems, and that takes time. Because of the shear size of it, the record ended up being 700 hours of mixing in 51 days. So there were a lot of really long days.

How long would it take to mix a typical song?

The first few songs took longer because you want to find the ultimate method and you're still getting vocal sounds set up and routing guitars and things like that. After a while the positioning and the numbers on the console start to get hammered into place, so it gets faster.

Some of the more complex songs actually went faster than some of much simpler songs, which you wouldn't expect. It feels done when the vibe is correct, and this record in particular very story oriented, so the energy level had to match where it is in the story. We actually started by mixing a few core tracks, and then after that we determined that we had to start at the beginning so that the mixes matched where they were in the story. Sometimes we'd be a few hours in and decide that we had to stop and change direction. That would take a bit of time.

Considering that the arrangements were quite large, did you find yourself muting any elements during the mix? Many times you can hear it in your head and it works, but when you finally hear the mix and have to match it to a story you find you have to pull things out.

That's a really good question. The short answer is yes. The tracks we got from David Campbell used a standard Decca Tree for strings, orchestra and brass and choir. In one of the choirs there was one soprano that was "iron lung." It was unbelievable how loud she was. Great tone, but she would just tear it up. In the Decca it was so overbearing that we had to resort to some of the spot mics to even out the choir. We'd have to pull down some of the elements as we got closer and closer to her because she was just so loud in the room. The choir wasn't recorded discretely, it was recorded as a group, so we had to go through and fine-tune elements which sometimes involved turning some stuff off.

What was the most fun?

The most fun was that as it got put together you could see that it was going to work. The fun was watching it all come together. It was such a huge undertaking that it fights the current trend of making things small. The band went the other way and said, “We’re going to make this big.”

For the last record, someone asked Jon what it was like to work with me and one of the highest compliments was that he said I know my stuff from a technical point of view but he said how much fun we had. Having fun in the studio affects the creativity. You only have a finite amount of energy, and the more you dedicate to negative things, the less you have to create. I try to carry that to all the projects I work on. But really, the fun aspect is that we get to make records for a living. We’re really blessed.

Billy Decker

Billy Decker has mixed hits for Rodney Atkins, Chris Young, Dustin Lynch, Jaime Lynne Spears, George Jones and Sam Hunt, among others, and combined they've sold millions of albums and streams into the billions. Billy is different from most other mixers in that he's able to mix extremely fast thanks to his templates that outline all the basic parameters needed to make each vocal and instrument work in the mix.

To that end, Billy isn't shy about how he does it and has recently released a book called Template Mixing And Mastering that explains his method. He's also helped to develop the Buss Glue series with Joey Sturgis Tones as well as his Drumshotz library with Drum Forge. The following is an excerpt from Billy's appearance on Episode 342 of my Inner Circle Podcast (he also appeared on Episode 192 as well).

Template mixing is something that you do and you're obviously great it, but wasn't Chris Lord-Alge the first guy to do that?

I'm sure he probably did on a large format console, to be honest with you. I have met him a time or two and he gave me a lot of really good pertinent advice. I really just wanted to mix and not spend 12 hours a day in the studio, so I started doing this template thing and it allowed me to almost have like a virtual assistant.

I've never had a real assistant. I prep all my own sessions and I've always just been kind of a one man show. By me doing that [mixing with a template], it allowed me to just create a skin per se, import the

audio in, and then I'm off and running. The song at that point is almost 90% to 95% of the way there. I realized that if I used gain structure and made the waveforms a certain height, they would fit in the template. In actuality, I mix almost as much with my eyes setting up my template as I do with my ears.

I've heard the story about Tom Dowd doing a mix very much like that, where by just using the VU meters he knew exactly where to put everything. The story goes that he could put up a whole mix without even listening to it. When he finally opened it up, it was like, "Wow, this is almost finished," so even way back then it was happening.

Well, the fun thing about it is I actually gave my template and my samples to three of my good engineering friends here in Nashville. I said, "Do your best to copy me." The cool thing about it is it's just a starting spot in my estimation, because halfway through the mix they decided to go this way when I would go the other way.

So they built it up, got the foundation, and then from there they sent me back three mixes that were totally different from what I would have done. All three of them were like, "Wow, this got up to a certain point really fast, and then all we needed to do was to make our rides and tailor it towards our ears and we're there."

It was designed to be predominantly a speed function just to get me home so I can be a dad. My wife was always like, "You're working too much in the studio. You're missing your kids growing up," and I really was. So by me figuring out this template thing, it really helped me to get my life in balance, as far as work and home lives. Every time something is in balance, everything is good, you know what I mean?

Let's go back to the template for a second. How long did it take you to develop that?

I've actually been modifying it ever since I started doing it probably around 1998. I had to do it because in my early days in Nashville, I came up in the demo world. I was given five to nine, sometimes 10 songs a day that were songwriter demos with a full band. They had to be completed for the next day so my songwriter, friends wouldn't miss a pitch and wouldn't miss out on like a Brooks and Dunn or a George Strait cut. It was up to me to make it sound good and get it done.

Fast forward 20 years and now it still only takes me about 45 minutes to mix a song. Whether I'm doing a record or whether I'm doing a demo or an independent project, I use all the same sounds and I do everything the same. It's just that I've learned how to maximize my time. Now it's funny - if I spend more than an hour on a mix, I'll screw it up. I'll start going backwards and I'll second-think myself. I've got the template dialed is so well that I actually have different templates for different types of music. I kept all the song templates that I think are appropriate over the years that work and I can just grab them. I guess the next phase is I'll start selling these templates [laughs].

At the level you're at, you're probably getting really good sounding tracks in so a lot of that hassle has already been dealt with.

Not true. You would be shocked. Sometimes I'll get record quality stuff done in the finest studios with the finest microphones, and other times I'll get home recordings. I'm working on a project right now where the drum bleed is so bad that my triggers on the snare are firing when the kicks go and I can't do anything other than go in and delete between every single snare hit manually. So I do have the luxury of getting very well recorded stuff, but on the other hand, I do so much independent stuff, and I still do a lot of demos.

My slogan is "Have Audio - Will Mix." I just love to do it. I always tell everybody that as long as it's not distorted I'll be able to get it to where it competes with anything out there. That's my goal on everything that I do, but I do get some stuff that's just a train wreck.

When I talk to other mixers, one of their biggest problems is the fact that they're getting tracks that are recorded by musicians who aren't very good at it, and then that leaves the mixer pulling their hair out trying to make it sound good. The fact that you're triggering things, especially kick and snare, kind of eliminates some of that.

Yeah. The overheads and the rooms do make a big difference on the sound of the snare drum because of the bleed though. If I get some really bad sounding overheads I'll just high-pass them up to 350Hz or so, then I'll use the triggers and I always blend in the real snare. Kicks from day one I have just triggered a hundred percent, no real at all. Snares and toms I will blend in about 50%

On press rolls on the snare, I always have the real one sitting here with maybe the samples leading the party over there. When it comes to a part where the dude rolled into a fill, I'll bring up the real so you get that and then I'll ride it back down. Everybody is like, "Wow, he made that work using his triggers. How did he do that?" Actually, all I did was just bring up the real snare [during that part].

One of the other hassles that many mixers have is the fact that the more independent the artist, the worst shape the tracks are in. That means that you have to spend a fair amount of time doing some prep on them, which is time consuming. Your actual mix may take 45 minutes, but how long is the prep time before that?

Most musicians these days are actually fairly competent engineers. Almost every musician has a home studio (even drummers) and they're doing overdubs at home. Some of those drum sounds actually sound better than what I've gotten in from major recording studios. I clean all my own tracks. I trigger all my own drums. I really don't even think about it. So for instance, I'll go through and clean a vocal on the fly. I'll hit play, and while I'm mixing, I'm actually cleaning up the dead space between the vocal and getting the breaths. I'll hit play and by the time the song's done the vocal is clean as a whistle. So that's three minutes tacked on to the 45 minutes. I guess it's 48 a mix. Sorry, I lied [laughs].

Is there a problem with the fact that your settings are mostly the same all with time? Is there a Billy Decker sound?

Yeah. Without sounding egotistical, I would like to think so. If you listen to any of the stuff I've mixed there is a similarity just because I love samples. I know Nashville's kinda in a love/hate relationship with samples. When I came to town in the late '90s, everything was kicks, snare and tom samples. Then it went to a real organic sound. Then it went back samples. I think a lot of people have accepted samples, and I'm using them now more as a supplement rather than running extremely sample heavy.

I try to keep the realism where before wouldn't even use the real snare. I lean towards a real clicky kick drum. A lot of times it's too clicky for Nashville, and they're like, "Dude, what are you doing? It's not a Metallica record. We're doing Texas country."

I always liked to use reverb on the drums or reverb samples, but I've figured out through the years that I need to make the decay stop before the next drum hit. So I'm all about timing the decay of the sample or the actual reverb to get out of the way and make the drums a little

more impactful where you can actually make them bigger by using smaller well-timed verbs. That's what I'm going for.

You're big for parallel compression.

Yes, on kick, snare and toms. I never run overheads or cymbals or room's or hi-hats through it. I know a lot of people do but I've just never been able to get it to work. Predominantly that's the only thing that I use parallel compression on. Everything else goes to my two buss [the master buss], other than my background vocals. I do run those through a subgroup wet and dry, so I guess that would be considered parallel-compressed. I also send it to that to the 2buss [master mix buss]. A lot of times I'll send it to both places just to get it above the track.

I noticed that you treat all keyboards the same.

Yeah, predominantly I will. All the keyboards I get are always in the same frequency range, so I'm always going to treat them in the same way, then the only thing that will differentiate them would be the source material. That's kind of my theory on electric guitars too. I always add just a little bit of that 7K, a little bump at 240, and a little bump at 100 on that API EQ.

Like you mentioned, I have a series of plug-ins that are endorsed with my name on them through Joey Sturgis (JST plugins) that I've actually built those settings into so all you do is put the plug-in on. It is already pre-EQed, pre-clipped, and pre-limited, and it's exactly those settings.

How do you deal with multiple electric guitars in the same frequency range and playing in the same register that sometimes clash? If you use the same settings on them, you're not really separating them.

Every once in a while, I'll go in and actually bump like some 3k or 4k to get it to bite through. Sometimes I'll put it in a little verb. Usually I run electric guitars just bone dry and panned hard left and right. I'm big on left or right or right in the middle and that's it. And then let the source material differentiate the frequency spectrums rather than just try to find a little hole for everything. I'm also not opposed to throwing stuff away.

I spoke at a summit for the Unstoppable Recording Machine, and I was on a panel up on a stage with a huge screen linked to my computer so everyone could see what I was doing. I was mixing and they were watching and asking questions. Then the drums came up and I'm like, "A sub-kick. All right, let's throw that in the garbage. We don't need that. Mono rim, no. Crush mic, no. Bottom snare, no. We don't need that because it's rattling too much." And I was just throwing out stuff left and right, and everybody was like, "Ohhhh."

Well again, mixers complain all the time that they are sometimes getting more than 100 tracks in a session, which is okay if they're labeled and you can easily decide what to throw away. The big problem is if they're not labeled well and you have to listen to them all and then try to figure out what to do. That takes a lot of time.

In Nashville we get a lot of electric guitars miked up with the SM57 and then a condenser mic as well. 9 times out of 10 I throw everything away except that 57. I'll get my three guitar passes in but there are six tracks, so I'll instantly just make those the 57 left or right. A little bit over here and maybe put a little delay on it to differentiate it, go. Turn it up or down and rock and roll.

You treat the vocal specially. Can you talk about that?

As far as the template goes, a male vocal is always going to sound like a male vocal and a female vocalist is always going to be within a certain range. The only thing that will change is the amount I'll pull out in the low mids. 9 times out of 10, the upper mids and the very highs I leave flat. I'll maybe add a little 15k, but I'm more of a cutter than an adder on vocals because I think vocals sound good the way that they are.

The thing is, when they marry up with the wrong microphone they may get a cloudiness. By just carving out a little of that low end it seems to clear them up. Then just put a little air or a sparkle on the top and smash it and double de-ess it and you're done.

I saw you use two de-essers because there's always more than one frequency that's a problem on a vocal.

Oh yeah. And one I always put in at 2k I call it the "Trisha Yearwood frequency" or the "Carrie Underwood frequency." That's not a knock on either one of those girls, but it's just that the 2k nasal thing that you can hear in their voice. If anybody goes, "Hey, can you knock down some of that?" I'm like, "Oh yeah, I know exactly where that is." So the de-essers seem to do a real good job and I always put them at the end [of the signal chain] because all of the compression and the limiting up top the processing chain always brings that up so it needs to get knocked back a bit.

Let's talk about your branded plugins. Are they available now?

They are. I developed them with Joey Sturgis at JST. He was like, "You and I are so similar in our approach to the way we work. The only differences is our source material. I'm a metal head and you're a country guy. I would love to do a series of plugins around your settings because they're so similar to mine. I totally get what you do."

How about your drum sample packages. Are they are emulating what you use?

They're called Drumshotz. I decided to actually take the exact combinations from the number one records that I'd mixed, using my exact samples that I used to make those records minus the real [the original drum sound], because the record company owns that. They're pre-blended and pre-everything, and balanced with all of my processing on it. You can throw them in Trigger or whatever you're using to fire off your samples and you don't have to do anything. Just put them in there and you are done.

Some of the fun of mixing is finding new plugins and finding things that you didn't quite expect that work maybe in a better way, but you're restricted from that because of your template.

Yes and no. What I do enjoy is coming in on my day off and practicing. I love trying new sample combinations and maybe finding a new way to do something, but I feel like I've got everything I need already. The wheel has been invented and I know it's going to roll.

I'm at the point where people go, "What is your favorite plugin?", I just point to my ear. If I can't get what I've got with what I have after doing it for this long, I need to just quit and start rebuilding motorcycles for a living [laughs].

What you do gets you a 90% out of the way there, but is there a favorite technique that you use that gets you the other 10%?

I think it's just listening and totally just going with my gut and not second-thinking it. I will crank it up in the studio and just, for lack of a better term, jam out to it. If it feels good, I'm done. I've learned after this long I can let it go, partially because I know my clients are going to have some input. I tell everybody, "I'll put you in the box and anything you want to do in there is subjective."

I'm just going to get it to where I know it feels good, then I'm going to bring in the producer or the artist or the songwriter. I'll let them play with it for as long as they want it. If it takes a whole day, then that means it actually did take a day, but I only spent the 45 minutes or so, let's say 48 or an hour like we said, getting it to where it felt good. I think what really makes cool records is the input from the artists, the writers, the producer, and just the little tiny things that make it kind of shine.

I love people coming over and hanging in the studio and contributing to their own music, so that's kind of the approach I take. The bring me their cake to "Decerate" it. I'll bake it and then let them frost it.

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DJ Swivel (Jordan Young)

Jordan Young, known to many as DJ Swivel, is a Grammy-winning producer, mixer and songwriter who's worked with a wide variety of hit-making artists that include Jay-Z, Diddy, Pharrell, Britney Spears, Beyonce, The Chainsmokers, BTS and many more. Jordan also has developed his own line of plugins based on his own unique processing techniques.

When you approach a mix, can you hear the final product in your head before you begin?

Yes, to a degree. Once you have the stems in front of you there's this element of discovery where maybe you're hearing things that you didn't hear in the demo, or seeing elements that you can leverage or improve upon. You might see elements that will provide challenges to where you need to go that you don't always hear when you listen to a demo. I really have to see what I'm working with, but yes, then I know where to go.

Where do you generally start a mix from?

Most of the mixes I do are contemporary pop, R&B, EDM, and even country pop. I usually start with drums because they're the driving force behind most modern music genres, especially hip-hop and EDM. I'm getting the drums to sit right and giving them the bounce that I want. Then I'll add the bass and get the drum and bass rhythm section going. Then I'll add the main instruments. I'm not adding any of the fluff or extra stuff yet. If it's a guitar-driven song, put that in. If it's a piano or synth-driven song, put that in and get that sitting well.

Then I'll usually bring the lead vocal in. I want to get that sitting right. After that I'll start adding in the background vocals and the other instruments and affects and risers and impacts. All that stuff comes at the end, but I have to get the meat and potatoes stuff first and it usually starts with drums and bass.

There's a difference in some of the music that you're mixing in where the kick drum sits. That's pretty genre-specific.

Yeah, 100 percent. And by the way, it's never set it and forget it. It's set it so my brain can focus on the next thing creatively. I might still go back and touch up the drums towards the end of the mix, but I've gotten it into a place where my brain can move on to the next thing. I always do a pass, then go back to the top and touch up everything again. Once you get all the instruments and vocals in, then the kick might have to come up.

That's why I said I generally know where I want a song to go but once all the stems are in and you start messing with things it sort of naturally evolves.

How much prep work do you have to do? Do you get tracks that have clicks and pops or are they generally clean?

Both. In some cases I get well-edited, well-produced stuff and in others I'm doing a ton of editing. I've found recently that I rarely have to do any tuning, where I used to have to do a ton of Melodyne. Artists are fickle with the way their voice sounds, so if they have an Autotune setting then I don't want them to send me the stems without it. If you got it and it's good, print your vocal stem with it. Get rid of the reverbs

and delays and any other things that you've added, but if you tune it, I have no problem with that. But every once in a while I do get stuff in with pops and clicks.

I do a ton of vocal editing. It's just my process, even if it's already been edited and it's good. Typically 50% of a mix I'm spending on vocals and most of that is on the lead vocal, especially when the hooks are all a little bit different. It's a tedious process because it's all clip gain automation and breath editing. Sometimes it's, "I don't like the sound of this breath. Let me find a better one that's a little smoother." I'll also do time shifting with little nudges, around 5 milliseconds here and 10 milliseconds there if I feel it helps with the flow of the song.

Part of the clip gain automation that I do is leveling out a vocal so it's consistent, but part of it is also building in performance. On a pop record, if there's a long tail on a vocal I will often create a crescendo at the end of the vocal so it leads into the next line. That's all about making the vocal sound dynamic and interesting and natural. Oftentimes with singers the emphasis will be on the wrong word of the phrase, so I'll turn the one up so it's more on the beat and turn the other one down so that it flows the correct way. I do a significant amount of that. It doesn't matter, I could have a near perfect vocal and I'm still going to do some of that stuff to touch it up. I've never had a song where I didn't have to do it.

Don't you get push back from artists for that?

No, because it's never any heavy strokes. It's subtle things that you feel and don't necessarily hear. It's like surfing and you're riding a wave. It's up, it's down, it's left, it's right, but in the end you still want to get to the other side of the wave. I view a song like that. It doesn't have to

be perfect, it just has to feel natural. That's where a lot of the taste part of mixing comes in.

When I'm working on vocals, the compression, EQ and de-essing comes pretty quick. I know where I want it to be and where I want it to sit. I hear the frequencies I don't like and enhance the ones I do.

And by the way, sometimes I'll get a session and I'll see all these channel strips with ten EQ curves. My stuff is really simple when it comes to that. I get rid of what I don't like. I do a high-pass filter on it so that there's no unwanted low end. Usually I'll add a little brightness on the top in pop stuff, but at the right frequency so it's not too harsh. The sound of the vocal generally comes naturally and it's not from a lot of EQ. Now I'm using Soothe and that's been very helpful with those upper-mids. I used to use sometimes three de-essers on a lead vocal but I wasn't necessarily using them as de-essers.

For female vocals, somewhere between that 2k and 4k range is super annoying. If you cut it out it often dampens the whole performance. Oftentimes I would stack one or two de-essers, just targeting narrow frequencies and setting it to where it would get rid of it only when the vocal got too intense in a louder more dynamic section when they go up in their register.

That's where I do a lot of the finessing, but the rest of it is just like drawing a picture. All the automation and the clip gain and the editing and the breaths, it's like taking a vocal and repainting it in a way.

You’re using Pro Tools, right?

Yeah. I used to be in Logic for my productions but now everything is in Pro Tools. Doing everything in one DAW makes life so much easier. I’m seeing a lot of producers that are not using Pro Tools though. Obviously Ableton is huge, Logic is huge and FL Studio is huge.

One of the challenges most recently with hip-hop, and every so often with EDM, is that they’re happier with the demo than they were before. In the last three years I’ve noticed that it’s more “touch up the demo.” They’re looking for the one hour mix and not the six hour mix that I used to do. They rock with the demo and are not up for quite as much interpretation.

When I got my first professional mixing job in 2007 with Fabolous, artists trusted the engineer because I think because the craft of mixing was more mysterious than it is now. Now there’s just content about it everywhere. You have producers who are more capable, and they get it to a place where there isn’t a lot of room for a mixer to interpret because artists hear it a certain way and they don’t really want to change their mind.

Do you have a method for setting levels?

Not really. When I start a mix I get my stems and they’re all at zero. In some cases the producer gives me stems where the gain staging is lower so I already have headroom. In other cases I’m getting stems where it’s sitting at zero so I’ll add an All group and bring everything down about 10dB. That gives me enough room to work. When you start a mix it never gets quieter - it always gets louder [laughs]. That’s where I usually start and very rarely do I have to readjust during the mix.

Do you have an approach to EQing?

For me it's a lot of the 2k and 4k stuff that I don't really love. That's also where a lot of the energy of the song is, so you can't get rid of too much of it. Coming from a hip-hop background, I love pushing 60Hz. I like kick drums with some nice 60 in there. If it's not the kick then certainly the bass. I like punchy kicks too so I'm usually looking for ways to enhance that so you feel it in your chest. Sometimes it's at 120Hz, sometimes it's at 150Hz

The pop background in me makes me tend to favor very polished mixes so adding 12k in a subtle way makes it nice and shiny

Do you use a lot of compression?

Yes, I use a lot but I'm not putting a compressor on every track. A lot of tracks have no plugins on them at all. Nowadays producers know what they're doing so a lot of times I get tracks where I don't have to do anything.

With drums, a lot of times you're getting pre-mixed samples from Splice and these other environments so you don't really need to compress the kick drum because it's probably been compressed four times by the time you get it [laughs]. I just try to be very pragmatic about it. If I feel I want a little more snappiness or a little more attack, I'll add some compression on the drums.

With the vocals it's really more about control. Usually I'll use two compressors and they're both set to be more subtle than using one aggressive one. I'll also do clip gain automation so you're managing the level going into the compressors, then volume automation post-compressor. In that regard, there might be four stages of level control on a vocal, but there are plenty of tracks in a typical mix that have nothing on them.

What processing do you use on your mix buss?

It varies. I've created a mix template that has all my settings when I start. Every session starts with this template. I have it in Pro Tools but I've translated it into Logic, Ableton, and FL Studio. I have them online and anybody can get them from my website.

On the master buss I usually start with the Slate FGX and I'll play with it until I get 1 to 1½dB of compression. Then I'll play with the high and low details on the plugin to see if I can get more punch or detail. After that I'll funnel it into Ozone. In a hip-hop workflow I'll have one process and in pop workflows I'll have another process. The hip-hop one I basically do nothing to eliminate any low end anywhere. In the pop process, there's often some very subtle filtering of the extreme lows under 30Hz or so. I'll also do some spectral shaping of the very lows at 40Hz and below. That's just catching things when they're a little bit too loud.

My Ozone plugin will have a multiband compression layer that's just kissing each of the four bands a little. I then use a little bit of general EQing and that varies. Sometimes it's getting rid of the 500Hz boxy stuff and sometimes just doing a very wide bell curve on the high end just to give it a little bit more shine. We're talking like a half-dB. A dB would be like an absolute maximum.

Sometimes I'll do some imaging stuff. That's actually one of the plugins that I created. Spread is a multiband stereo imager. That came from the processing that I'll typically do on pop songs. I'll usually fan out the frequencies where the highest frequencies will be the widest, and the upper mids a little less wide and the lower mids even less than that. It's very subtle when you A-B it. I don't believe that anything on the mastering side should have broad brush strokes. If the mix is right you shouldn't have to do anything crazy on the master buss.

Tell me more about your plugins.

We have four plugins, some sample packs and some bundles at DJSwivel.com/products. The first plugin called The Sauce launched just before the pandemic started in January 2020. It's a vocal chop tool that has pitch and formant shifting, saturation, distortion, chorus, flanging, stereo imaging, a delay, three different types of reverbs and output filters. That's a multiband function where you can change the pitch and formant of each band independently, which I think is very unique.

The concept behind that came from when I was working a lot with The Chainsmokers and BTS and I was creating these vocal chops where I found that I'd have to daisy-chain like seven plugins to get what I wanted. I just thought that it would be easier putting it all into one plugin with presets that quickly get you where you want to go.

Shortly after that we came out with Spread, which improves on the imager in The Sauce. It's by far our most used product with over 100,000 users. After that we came out with BDE, which is our flagship distortion plugin. I wanted a distortion that did not impact the

dynamics of a sound so we created something called “Dynamic Preservation” that keeps the dynamics intact.

The fourth plugin that we just came out with is called Knocktonal, which is a resonance enhancer. Rather than using general frequencies, we use the frequencies and harmonics of notes to reshape your drums in the key of the song, almost like drum tuning without actually destructively tuning it by pitching it up or down. You’re just enhancing the resonant frequencies that you like, or getting rid of resonant frequencies you don’t like.

What monitors do you use?

I’ve tried all the hot monitors out there but I keep coming back to Genelecs. I started on Genelec 1031s and 1035s. Before I moved into my current house with a proper studio, I mixed in a living room in my apartment in North Hollywood next to a noisy refrigerator. It was a terrible listening environment, but the speakers (8351s, 8361s, and 8320s) and the built-in room tuning really made a noticeable difference.

How many revisions of a mix are you asked to do?

It varies. Usually it’s around five or six but I have an album now where one of the songs got 15. I don’t limit the number of revisions because in my view an artist has to be happy. As long as we’re moving forward I’m never going to get annoyed. When we start mixing in circles and it’s clear that the artist or producer is just indecisive, then I’ll say, “We have one this way and we have one this way. I’m not doing another one.” If an artist comes into the mixing process unsure of themselves and doesn’t want to trust your vision, it’s just tough to get to the result that we all want. That’s rare when that occurs, and it’s usually never with major artists.

How many versions are you asked to deliver?

Usually I'll do the main mix, acapella, TV mix, clean if I need to, and that means a clean acapella and clean TV mix too. I'll also add mix stems as well.

In some cases I'm also asked to master. I do not purport myself to be a mastering engineer. I can do it and I've had some success with it, but it's not what I do and I'd rather send it out for mastering.

What's one big thing you'd suggest to someone who's just getting started as a mixer?

This is an exercise that I used to do. Take a piece of paper and listen to one of your favorite songs and write down every single thing you notice in the song as best as you can. What's the genre? What's the tempo? What's the drum rhythm like? What are the sounds like? What instruments do you hear? What vocals do you hear? How are they stacked? Do you hear a harmony? How are they panned? What's the reverb or delay like? Are there any mistakes in the mix? Are there things you would have done differently?

When you ask a new producer or engineer to do that, their list is usually pretty short. They name a couple of things and then stop. As they get better and do it more and more often, they will start to hear more of the nuances and the list will get longer.

Do that every day for six months. It shouldn't take more than 15 minutes. Then in six months go back to that first song you did. Your list

is going to be three times as long. The best thing is, it’s easy to see your growth. It really works in developing your ears.

Jimmy Douglass

After learning at the knee of engineer/producer Tom Dowd during Atlantic Records' glory days, four-time Grammy winner Jimmy Douglass (affectionately known as "the Senator") has gone on to become one of the most sought-after engineer/mixers in R&B, hip-hop, and rock. One of the few engineers who can cross genres with total ease and credibility, Jimmy has done records for artists as varied as Otis Redding, the Rolling Stones, Foreigner, Hall & Oates, Roxy Music, Rob Thomas, Snoop Dogg, Jay-Z, the Roots, Ludacris, Justin Timberlake, Timbaland, and Missy Elliott. But having old-school roots doesn't get in the way of Jimmy working in the modern world, as you'll see. You can read more about Jimmy and his Magic Mix Room at jimmydouglass.com.

How have things changed between when you first started and now in terms of your approach to mixing?

The urgency factor has definitely disappeared. Now we don't use a lot of musicians in the stuff that I do, we use machines. Everything is totally replaceable. As a matter of fact, you can erase a part that somebody played, and they'll just replace the part and nobody seems to care anymore. Back in the day, it was a major deal to replace anything.

And the rough-mix thing is becoming the nemesis of all of us now. Record companies want change and yet they don't want change. They want it to sound like the rough, but they want it to sound different. Someone will hand in a rough to a record company after taking a lot of time to make it sound good; then they'll hand it to me to do what I do. When I do what I do, they'll say, "Oh, it doesn't sound like the rough," and I'll think, "How am I going to beat a rough that somebody worked

on for a month in six or seven hours?" As a result, I've been starting to match the rough. I never used to listen to them because then I couldn't really do what I do. Now it's the opposite. If you don't get close to the rough, the mix will probably never be accepted.

How long does it take you to do a mix?

It's beginning to change a little bit, but I'm a basic 10- or 12-hour man. Back in the day I could mix four or five songs in a day, but I just don't know how to do that anymore. Then again, back then you recorded what you wanted to hear in the end. Now people want to imagine things they don't hear.

One of the big things is that we might only actually spend maybe 4 hours of the 12 mixing the record because there are so many visitors and interruptions. People think nothing of stopping your mix and taking the time to play a whole record for a friend. I was in the groove, now I'm not in the groove anymore, and it takes some time to get back into it. We used to listen to records to get ideas or to try to emulate, but you were always working the whole time you were there. Now we might end up staying until 5 or 6 a.m., when we could've been done at like 1 in the afternoon. [Laughs]

Also, there are so many people hanging around or coming around to listen. Back in the day, the only people hanging around in the studio were part of the band or had a really good reason to be there. Now there are people who aren't connected to the project who are giving their opinion and who aren't really qualified to give an opinion.

Is mixing rap different from R&B or rock?

The tracking is so generic and sequenced and simple that the tracks have no real harmonics or overtones. There's nothing that's different, so it's really kind of simple. A lot of times I'll even use a stereo mix that the producer gave me because they can't find the original session to break the individual parts out, so all you're really doing is just putting the vocal on top. You have to try to make something sound really special out of something that's not.

That said, I've been doing a lot of EDM [electronic dance music] lately, and I really like it a lot. It has me excited and learning new stuff. As long as that happens, I know I'll be doing a good job.

Since you do all sorts of music, from rock to R&B to EDM, is your approach the same or do you prepare differently depending on the project?

The one thing that I do is something I call "tuning my ears." I listen to a lot of stuff in that particular genre to get to know what the particular sound of the day is. You want to sound contemporary and current, but you can't know what that is unless you listen to the records that the audience is digging at the moment. I'm not saying to copy it, but I tune my ears to know what the parameters are, so I listen to the genre to go, "Let's see what's considered cool today."

With some old-school guys, they're still making the same kind of records; but I'm making young records, and they're being made totally different from the way we used to do it. All the things they're doing I identify with because I was there, but they don't exist anymore.

I approach mixing records like fashion. This week tweed might be in, so even if I'm giving you the best silk in the world, you're not going to be

interested.

Speaking of which, how much do you mix in the box?

I'm living in the box because I'm getting mixes that everyone wants recalls on. [Laughs] That's all anybody really cares about these days. "Give me a recall. I want to change something right now" is the attitude. They don't care what it sounds like. I have that great big [Neve] VR down in Miami, but I'm not using it much anymore.

Do you mix with a mouse or use a controller?

I have an SSL AWS that I use when I'm mixing in the box. I like to grab faders. If I want to put my fingers on five vocal faders like I used to, it's the only way to do it. That's a dexterity and relationship that I have that people who mix purely in the box don't have. It's like playing an instrument in that it's just a feel thing. I'm sure it can be done in the box, but it doesn't feel the same. There's an eye/ear relationship to the distance of a fader's throw that's important.

Where do you control your automation from? Do you use faders or draw it in?

I'm trying to do as much as I can in the box, but when it comes down to vocal rides and things like that, I still have a better feel for using the faders rather than drawing in words. That works, but it feels so cold and calculated to me. Vocals have always been the most important part of a record to me anyway, so the way I control them has to feel right.

Do you have any tricks to make things sound good in the box?

I use an outboard summing amp for the warmth factor. I also use inserts with actual analog gear, but I always print it. If I use some effects in the analog domain or use the EQs off the board, I'll print it right back into the box. Sometimes the plugins just don't do what I want, so now that's what I'll do so I can still do recalls later.

Do you have any go-to plugins?

I have a lot of plugs only because I collect them all. I have two theories about how to use them. On some days I'll go to specific ones because I've had them for a while and I'm used to them; then on other days I think, "What's the difference? A digital EQ is a digital EQ." I go between days of hearing the difference and then not.

Do you use a lot of effects?

I'm from the old school where if you recorded it right, there's nothing to really add. I don't mix like that now, of course, because people want more bells and whistles and stuff. I always approach it like, "How can I give you what you put in here without changing it?"

I'm still using some of my analog effects because they do sound different. They don't have that harsh, cold, bright digital sound, especially flangers. The whole thing about a flanger is that it's nice and warm because the top end is cut off, but all you get with digital is top end. The same thing with digital delays. And I still use tape machines for slap- back. That also acts as my buss compression, since I always mix to a tape machine.

Are you compressing things more than you used to?

Absolutely. Everyone seems to want it smashed in your face now. I'm not saying it's a great thing, but I'm just doing what people want to hear. Buss compression has never been my thing, so my innate mixes usually sound lower than everyone else's right off the shelf. I put an [Waves] L2 or an L3 on it just to make it sound as loud as everything else, but I take it off when it goes to mastering. That lets the mastering engineer do his thing and bring the level up to where it needs to be.

How much EQ do you use? Are you old-school about it?

I use a lot of EQ. I like to bring the sound to me. I also add EQ, I don't subtract it. I know most of the world doesn't do that, but that's how I learned. Since I didn't know anything, the only way I could tell what an EQ did was to turn it to where it brought something to me. Most engineers try to get something out of the way so they can hear other things.

That's really cool, but I don't know how to do that. If I'm listening to a vocal and I don't hear it in my face enough, I bring it out to me.

Do you replace drums or just add to them?

I never really did drum replacement much. There are so many samples out there that I figure that the producer already got the sounds exactly as he wanted it. If I have to replace all the sounds, then what is the producer doing in the first place? That's one of the great things about [producer] Timbaland. He has the innate ability to put together sounds that work. He can hear what instrument goes with what, but so many producers have no ear for that at all. I always try to remain as close to the original product as I can, because the people that created it would've already changed it if that's what they wanted. Besides, everyone has access to all the same toys and samples that I have. Since

people have been living with a rough mix for two or three months with those particular sounds, if you change something they'll freak out on you.

Do you have to do a lot of fixes when you get a mix?

I've never been a guy who's done much of that. If you listen to my records you'll find that there's crap all through them. You can't do it all, so what's most important: the vibe of the record or the clicks and pops? I've always been a vibe-of-the-record man, so on my records there may be some things that others might find unacceptable that just don't bother me or my clients.

I do have assistants who rename and reorder the files for me before I start so I don't have to look at a whole bunch of files whose names mean nothing and have to figure out what they are. One of the biggest parts of the new mixer's world is having that support system around you. Without that support system, you can't compete.

That said, with the way things have changed, it comes to me at least sounding pretty decent to begin with.

What monitors do you use?

I'm still using my NS-10s, although I also have set of KRKs and a set of JBLs LSR 4300s with a sub that's ridiculous. How many versions of a mix do you do?

I still supply a vocal up and a vocal down, an a cappella, and lead a cappella, and an instrumental, and then I'll create stems for them so they [the client] can re-create any other combination for themselves later if they want.

The digital recording revolution is a great triumph for audio. It brings such great flexibility and versatility to the craft, but it also brings the element of mediocrity in the creativity part of the art. No one has to make any decisions now, and consequently, nobody really knows or thinks about what they really want. It's digital, they think, so you can always change it later.

Benny Faccone

Engineer Benny Faccone is unique in that he's Canadian from Montreal, but most of what he works on is Latin. From Luis Miguel to Ricky Martin to the Latin superstar rock band Mana to Santana to the Spanish remixes for Boys II Men, Toni Braxton, and Sting, Benny's 10-time Grammy-winning work is heard far and wide around the Latin world.

What's the difference between doing a song in Spanish and one in English?

First of all, the way they sing in Spanish is totally different than English. The syllables don't fit the same way. With English music, it feels like the voice fits right into the music rhythmically. That doesn't happen with Spanish singing because it has different accents with harder S's. You have to treat it a different way on the mixing side by building the rhythm track around it. It's a different flavor with a different kind of emotion.

Are there any other differences between doing an American record and a Latin one?

Everything I do is treated like an American record. The two have gotten a lot closer in terms of the music, but they've also become way different because Latin artists are so concerned with their vocal level. The vocals have gotten so loud on the Latin mixes that it borders on ridiculous sometimes.

Do you just do Latin pop or do you do any traditional salsa?

As a matter of fact, I do everything. The Latin field is not very specific like the American market, where you do one type of thing and that's all you do. In Latin music, you just do it all. I've even done a couple of mariachi records. There were a few records where they wanted some traditional salsa, and the only way to get it was to go to Puerto Rico and do it there. I had to get some ideas of how to do it from some engineers down there since they have very specific placement for a lot of the instruments.

And what is that exactly?

They've got two or three different ways of doing it, but the things that stay the same are that the shaker and the bongos are always in the middle.

Do you have a philosophy or an approach to mixing?

The only approach is to try to figure out the direction of the song, develop a groove, and build it like a house. It's almost like a musician who picks up a guitar and tries to play. He may have the chart in front of him, but soon he has to go beyond the notes in order to get creative. It's the same thing with mixing. It's not just a thing of setting levels anymore, but more about trying to get the energy of the song across. Anybody can make the bass and the drums even out.

How do you build your mix?

It really is like building a house. You've got to get the foundation of bass and drums and then whatever the most important part of the song is, like the vocalist, and you've got to build around that. I put the bass up first, almost like the foundation part, then the kick in combination with the bass to get the bottom.

Sometimes you can have a really thin kick by itself, but when you put the bass with it, it now seems to have enough bottom because the bass has more bottom end. I build the drums on top of that. After I do the bass and drums, then I get the vocal up and then build everything from there. A lot of mixers just put the music up first, but as soon as you put the vocal up, the levels become totally different. After all the elements are in, I spend maybe a couple of hours just listening to the song like an average listener would, and I keep making improvements.

Do you have a method for setting levels?

Yeah, I have a starting point. I usually start with the bass at about –5 VU and the kick at about –5 [on an analog console]. The combination of the two, if it's right, should hit about –3 or so. By the time the whole song gets put together and I've used the automation to adjust levels, I've trimmed everything back somewhat. The bass could be hitting –7 if I solo it after it's all done.

Do you put the snare at about the same level as the kick?

No, that's more a question of feel than level, because it has so many transients that it could be reading –10 and still be too loud.

What's your approach to EQ? Do you have certain frequencies that you always come back to on certain instruments?

Yeah, as a starting point, but I'll do whatever it takes, depending on how it was recorded. For bass I use a combination of a low frequency, usually about 50Hz, with a limiter so it'll stay tight but still give it the

big bottom. Add a little 7k if you want a bit of the string sound, and between 1.5k and 3k to give it some snap.

For the kick, I like to have bottom on that, too. I'll add a little at 100 and take some off at 400, depending on the sound. Sometimes I even take all the 400 out, which makes it very wide; then I'll add some point at 3k or 5K.

On the snare I give it some 10k on the top end for some snap and 125 Hz on the bottom to fill it out a little more.

For guitars, usually 1.5k gives it that present kind of sound. Pianos and keyboards vary so much that it all depends on how it feels in the track.

For vocals, it really depends if it's male or female. If they sing really low, I don't add as much bottom end. Usually I always take some off at about 20 Hz to get rid of any rumble, but anything on up, it really all depends on the singer. I might add a little bit in the 4k to 6k range as well.

How much has your approach to mixing changed since most of the recording world has switched to Pro Tools?

The major thing that's changed is the way I approach the bottom end. I never used to put a stereo limiter across the mix buss, and if I did, it was really light. Now you almost have to make it sound like a finished product and make it tighter sounding, so I don't spend as much time working on the bottom end as I once did.

I'm also concerned about how things translate to the radio more these days. Nobody's listening to CDs anymore, but they do listen in their car. I have to readjust how I do things because all the compression and EQ that radio uses is based on the 99 percent of people mixing in the box. When I hear my stuff on the radio, it's got more compression and it's fatter (mostly because I like it that way), so the station's compressor is hitting it harder, which means I have to readjust what I do so it works better on the radio.

What compressor are you using on the stereo buss?

I have an old SSL that I use 90 percent of the time. If I need something that's not as aggressive, I rent a Fairchild or something like that.

Do you begin your mix with the compressor across the buss?

No, after I have the foundation built, I put the buss compressor in. I hit it so it just barely moves the meter. With the SSL, the louder you crank the gain, the grainier the sound gets, so that's why I always try to start as loud as I can so I don't have to add more than 2 or 4 dB of level at the most.

What do you use in the box?

I use plugins for effects, but very few compressors or EQs. Only if I run out of outboard gear or stuff on the console will I resort to what's in the box. I do use more effects now because it's easier to do it in Pro Tools than hooking up the outboard gear. [Laughs] I like using flangers and Moogerfoogers and stuff like that to give it a little bit of something

different. If I can, I try not to insert it on the same channel so there's some stereo space to it.

Are you mixing to only Pro Tools?

In my studio [The Cavern] I am. If I mix at Conway or one of the other studios that I like to work at, I'll mix to 1/2-inch [tape], too. Pro Tools is winning out more and more instead of 1/2-inch tape because of the clarity. On something I did recently we mastered one song for seven or eight hours, going back and forth between the 1/2-inch and the Pro Tools file. The 1/2-inch had a warmth to it, while the Pro Tools file had a clarity. It was thinner but it was clearer. Neither one was better, but they were both so different. We ended up trying to add the warmth of the 1/2-inch to the Pro Tools master, but after trying different things, it still didn't work.

What's your approach to individual track compression?

Limit the heck out of everything. [Laughs] I like to compress everything just to keep it smooth and controlled, not to get rid of the dynamics.

Usually I use around a 4:1 ratio on pretty much everything I do. Sometimes on guitars I go to 8:1. On the kick and the snare I try not to hit it too hard because the snare really darkens up. It's more for control, to keep it consistent. On the bass, I hit it a little harder, just to push it up front a little more. Everything else is more for control rather than to have it stick right up in your face.

Do you have any special effects tricks that you use?

I use a lot of the old PCM 42s on guitars for a very short slap delay. It's mono, but it sounds really big. I use something like 4, 8, 11 milliseconds, so it doesn't sound like a delay. Sometimes I use as much as 28 milliseconds on a power guitar. You stereo it out, and it'll sound like two guitars on either side of the speakers.

Is there a certain listening level at which you always listen?

Yeah, I listen at a fairly modest level, not loud and not soft. When I start the mix, I crank it on the big speakers to kinda get hyped a little bit and check out the bottom end; then I'll slowly start listening softer and softer.

Does anything change for you when you get something to mix that you haven't recorded?

When I get a mix in, I keep some of the automation but take most of the effects off. It seems like nobody builds a song from the recording stage anymore. They throw a lot on, and you have to figure out what to do. That makes it really difficult because someone has been living with the song for months, listening to their version of the mix, then they give it to you and within a day you're expected to do whatever you want. It's almost like editing a book before reading it. What do I take out? If I take something out, how do I know if it's something that's needed somewhere else in the song? I almost don't know what to take out or put in until I finish the song, but you can't finish the song until you decide those things. It's a catch-22.

There's so much more wasted time now, because when they send you the session file, you have to prepare it. Since they're not in the room with you, after you send the mix out you have to wait for them to get back to you before you can do the tweaks. I have to give parameters because I'm in analog and I can't wait that long.

Do you do many alternate mixes?

I still provide alternate mixes so I don't have to recall anything. Ninety-nine percent of the time, recalls are not necessary anyway. The funny thing is that if you give them a vocal up, they'll use it. Give them a vocal even higher, and they'll use that. I still do it, but it's mainly for me for "just in case." I do not want to come back to remix. Once I'm done with a song, I've heard it so much that I don't want to hear it ever again.

Jon Gass

With more than 80 Top 20 hits, 100 Top 40 hits, and more than a hundred gold and platinum albums, Jon Gass has long been the go-to mixer for a credit list that reads like a Who's Who of music superstars, including Madonna, Whitney Houston, Janet Jackson, Celine Dion, Mariah Carey, Mary J. Blige, Usher, Babyface, Earth, Wind & Fire, Lionel Richie, John Mellencamp, and many more. Jon's unsurpassed style and technique have elevated him to a most esteemed position among engineers, working with the best of the best on some of the most creative and demanding music being made today. In this updated interview, Jon describes his method of mixing "in the box."

How much do you mix in the box?

Everything. I haven't touched a real fader in over seven years.

Do you use a controller?

Nope. It doesn't bother me to use a mouse to mix at all.

Can you describe what you do when you first get a track in to mix?

I get so many kinds of projects from all over the world, and some of it is recorded great and some of it is not. Frequently I'll get the stuff that no one else can figure out; the 150 unlabeled tracks kind of thing. Some of it probably couldn't be mixed in the analog world.

First of all I try to get everything organized. A lot of people recording today have never used a track sheet, so they don't understand organization. Their kick might be on track 80 and the snare on 18, so I have to get it to where

I can find everything first. Probably the most frustrating thing about having upwards of 150 tracks to deal with is the different mix pages [in the DAW] that you need to organize things, where on a console you could just reach for a channel and EQ it fast. I do miss that part of working on a desk.

After I know where stuff is, I try to get some basic levels and find out where the problems are. I don't do any fader rides, so if something's not working on a track or a section, I'll split it off onto another channel—that way, I can change the EQ and levels for different parts of the song as I need to. I've had stuff where I'd had to split the lead vocal onto seven or eight tracks so they could all be EQ'd differently because the comp track was from different studios recorded with different mics. Almost every word had a different timbre. It's maddening, and you can't get it to sit in a mix because it's like the frequencies are constantly changing. I still had to automate the EQs just to try to get it to sound similar all the way through. After five or six hours I can finally play through the song through without pulling my hair out. Luckily about half the stuff I do these days I'm producing, and I'm always cutting in mix mode, so it sounds like the record right from the beginning.

Do you still use any outboard gear?

I still do occasionally, mainly something like an Avalon 2055, because I've never found a better EQ. When I first started playing with my rig I couldn't even finish a mix because it just sounded like "the box." I started looking into analog summing and found the Dangerous Music

2-BUS. As soon as I got that, it totally changed everything in that it now sounded like the big analog mix that I was used to. The beauty is that a lot of the hardware effects that I use can now be blended outside of Pro Tools, so it sounds really analog.

Do you have any go-to plugins?

I use a lot of the stock Digi plugs like the basic EQs and compressors because I can use a couple hundred of them if I have to and still have some processing power left over. I like the Oxford EQ too, but for real serious work I'll still throw an Avalon on something. Some of the Waves Guitar 3 stompboxes I love to use as well, even though they're made primarily for guitar.

Are you printing your outboard EQ?

Some stuff I will print if I'm running out of outboard EQs.

Do you have a philosophy about what you're trying to accomplish?

Not really, I just go for it. I'm kind of a musical mixer. I think I try to find the more natural tones of instruments and maybe boost them in that direction as long as it all still fits together. I always think of it as a layer cake, so I just kind of layer the thing.

Can you hear the final product in your head before you start?

Actually, yeah, I can. I know some people push up just the drums and work on them for a while first, but I start with everything in the mix and work on it like that. The reason is that the vocal is going to be in

there sooner or later anyway, so you might as well know where it's sitting and what it's doing, and all the instruments are going to be there sooner or later, so you might as well just get used to it. I think that helps me see what I need to do within the first pass, so it doesn't take me long to get a handle on where the mix is heading.

How do you go about building your mix if you have everything in?

I really start searching out the frequencies that are clashing or rubbing against each other; then I work back toward the drums. I try to keep the whole picture in there most of the time as opposed to isolating things too much.

You don't solo things much, then?

Well, I do, but to solo something and EQ it is insane because it's not relative to anything. If there are two or three instruments that are clashing, that's probably where I get more into the solo if I need to hear the whole natural sound of the instrument. I'll try to go more that way with each instrument unless there's a couple that are really clashing; then I'll EQ more aggressively.

I always mix alone now, but when I used to have clients in the room with me I was always afraid to solo stuff in front for them because I think individually the tracks that I mix almost have to sound bad to work together. It really doesn't matter what it sounds like by itself, because it has to work together with everything else. That's where some of the young producers blow it. They go through and solo tracks and make everything sound fat; then when they put it all together they have a big car wreck.

You're doing mostly EQ cuts? You're not adding anything?

Yeah, I'm definitely into cutting more than adding.

How long do you think it takes you to do a mix?

I don't know that I've done any mix in less than two days. It's usually two to four. I won't take a one-day mix. I can't do it that way.

I usually try to get the rough mix so I know where the client's ears are at, but I don't feel like I'm getting anywhere until my mix begins to smoke it. Usually when I'm finished the rough sounds pretty small by comparison.

How do you use compression?

I try to make it so it doesn't sound like I'm using any. On certain things I'll stomp on it to make it sound compressed, but mostly I'll try to make it so you don't notice it. I don't compress the individual tracks much because if the stuff's EQ'd and layered right, you don't really need to do a ton of compression on the stereo buss. If the thing's laying right, at least with R&B it just kind of sits there.

Are you compressing the master buss inside the box or in analog?

There's some that I'll do in the box and then some outside as well. I have an SSL compressor and a couple of other things that I've used for years. I also compress some of the outputs going to the Dangerous box. It just sounds better that way.

Do you add your effects right from the beginning or do you wait until you have everything balanced and then add them?

I add them as I go. I hardly ever use long halls or long reverbs. I use a lot of effects that are usually set for tight spaces. Sometimes it doesn't sound like I'm using anything, but I might use 20 different reverbs, although not all are actually set for what you think of as reverb. I'm just trying to create more spaces. Though you may not hear it in the mix, you can feel it. If you take them off you'll really miss them, but you don't always notice that they're there.

When you're saying you use tight spaces, are you trying to move stuff back or just put it in its own space?

Yeah, put it in its own space. Sometimes it can be just a chorus, or even a harmonizer with a really short delay time. What it comes down to is I like short, dry sounds.

How short?

Like 25ms or less. I use a lot of 10, 12, 15ms on things. In the R&B stuff, you get a lot of stereo tracks that really aren't stereo. One of the first things I do is to widen the thing out, even if it's only 3, 5, or 10milliseconds, and just get that stuff separated so I can keep my center cleared out. I don't really like that "everything mono" thing.

So what you're trying to do is to make things bigger instead of pushing them back.

Yeah. I think part of that is probably from my early recording days. I didn't really have any reverbs, so I had to use more of the ambiance that was available. That started adding such a new twist as opposed to everything being miked so close and direct all the time. It adds such a great depth to everything.

Do you have an effects template?

If I'm doing an album, I'll try to use more or less the same effects for each song so there's some continuity, but otherwise not really. I do so many different genres and get stuff in from all over the world, so I have to add effects as I go because I'm not always sure where the track is going. I do miss the old days when I had 30 outboard effects plugged up at all times that I either used or didn't depending upon the song.

Are you timing your delays to the tempo of the track?

Depending on the song, yeah. Mainly the eighths, quarters, or sixteenths, but depending on the tune, I'll add in triplets or whatever feels right.

The other thing I like to do with delays is to diffuse them. I'll put a delay through a bunch of stuff just to make it sound worse. Sometimes I'll use Lo-Fi [the Pro Tools native plugin] or something like that just to clip the top and bottom end off and diffuse it off the lead vocal a little bit.

We joke about this guy that mixed a long time ago, and he'd have his delay clearer and brighter and louder than the actual lead vocal. I

think that's what kind of got me experimenting with ways to really tone it down.

With all the effects you're using, it sounds like there's a separate one for each instrument.

Absolutely. I very rarely use the same effect on more than one thing. How often to you replace drums?

I always try to use the drum sounds I'm given and then add what's missing to them. That way it's not completely different from the original drum sound. Drumagog is great for that. I have an 80-gig drive of samples that I've collected over the years.

Do you do a lot of revisions?

Not that many. I don't send a mix for the client to listen to until I'm totally happy with it. Ninety percent of the time there may be no changes at all after I'm finished, and if there are, they're pretty minimal. I just know that when I'm happy with it, they're usually pretty pleased.

What kind of alternative mixes do you normally deliver?

I usually only give them the main mix, the TV mix, and the instrumental.

What speakers are you using?

I mix pretty quietly on the mains, which are Augsbugers with a single 15 and a horn, but you can still feel the power and the punch. I just grew up with them, and I can't use any others. I still have the NS10s and Auratones that I use as well.

Do you have any listening tricks, such as going down the hall or out in the car?

I like to listen outside the room, but one of my favorite tricks is to turn on the vacuum cleaner and lay it up against the wall in the front of the room. Sounds a little strange, but I just kind of want to see if the mix is still cutting through at all. A blender works, too—making margaritas or something. [Laughs]

What percentage of time do you spend working on the mains?

Seventy-five percent or so. I listen pretty quietly, but when starting the mix I'll crank them pretty loud, too. At the end I'll flip around between the different speakers. I'll go really loud on the NS10s and do some adjusting; then

I'll go extremely loud on the big ones and do some more adjusting just to fine-tune things, but I mostly like it quiet on the big ones. I always said I could make a great sounding record on a cassette deck if I just had the right monitors.

Dan Korneff

Producer, mixer and engineer Dan Korneff has not only worked with prominent names like Breaking Benjamin, Paramore, Papa Roach, Lamb of God and My Chemical Romance, but has developed some of the coolest audio plugins available.

Dan not only knows how to expertly use his audio gear, but he also knows what makes it all tick inside as well, being very hands-on when it comes to modifying consoles, vintage outboard gear and microphones.

That led Dan to create his first plugin called the Pawn Shop Comp, which is like a combination of vintage tube and FET compressors blended together to create something really unique. He has since gone on to develop some very cool and unique plugins like the Talkback Limiter, the Amplified Instrument Processor, and Micro Digital Reverberator, all of which he uses on his mixes.

In the interview we talked about what he thinks is the most fun though, and that's mixing.

Do you have a specific approach to mixing?

That's a god question. Everything is sort of on autopilot. I know what I want to do and I know what I want it to sound like. I listen to the reference mix and get the idea of what they're trying to go for, and then make it my own.

Do you approach a mix differently if it's something that you tracked?

Absolutely, only because I already know what I want to do when I'm building up the song. Tracks that I get from other people, especially younger engineers and producers, I feel like I have to treat a lot differently. The sounds aren't baked into the tracks anymore. Everything is up in the air. On older records you can push up the faders and that sounded like a record already. These days it's more like, "This is the idea of what we want the song to be like. It's your job to figure it out."

Can you hear the finished product in your head before you start to mix?

As a mixer, yes. I can envision what I want it to be and start carving and tweaking until I get there. Sometimes when you're in the recording phase, not so much. Maybe you have an idea of what you want and you're kind of tinkering along the way, and then a couple months later you go, "Yeah, this is it." With mixing you don't have that luxury. You have to have a vision right away. If you don't have a vision it's going to take you a lot longer.

Some mixers will try to make every track heard regardless of how many there are while some have no problem with throwing tracks away to make things work. Which camp do you fall into?

I'm somewhere in between. When a band hires me, I feel that I work for them. It's my job to get their idea across with the way I do things. On the other hand, if you have dozens of tracks doing the same thing, it's your responsibility to say, "Hey, you don't need all this stuff." In that sense, you have to show them the difference. Try to make it all work because they had some kind of vision, but then when it's simplified you have something to focus on rather than a thousand things at once. Here's what it sounds like. 9 times out of 10 they'll

choose the simpler version, but it's your responsibility to show them that.

On tracks you get in to mix, how much editing and cleaning up do you have to do before you begin to mix?

Zero. As far as cleaning up, a lot of that is a creative choice. Maybe I'll strip out some noise that's on the toms or isolate cymbals, but if you want it tuned or edited, that's your job.

What DAW do you use?

My main DAW is Cubase. When I first started I was on Pro Tools like everyone else. Every studio had Pro Tools so if you wanted to be compatible you had to use that platform. I remember sitting there one day thinking, "There's got to be a better way to do this. If I have to sit here and wait for 45 minutes for crossfades to happen with Beat Detective. . ." That seemed silly to me so I went on this quest to find something better.

I tried everything that I possibly could. This was probably back in 1997 or so. Then I came across Cubase version 1.5. It had everything I needed, but I still couldn't make the full switch so I had two rigs and first. After spending a lot of time doing transfers to Cubase I just decided to stay there.

Where do you start your mix from?

I start with drums. I like to get that in the ballpark, add the bass, and then the rhythm guitars. That's my bed of music that I like to get

pumping, then I move on from there.

Do you have a method for setting levels?

I do. I'm still mixing old school. I still mix on an SSL. I put all of my faders at -14 all across the console, then start assigning channels from Cubase to the console. When I first start my mix everything is static at -14. I do all the balancing inside of Cubase. Once I start mixing I do all the automation on the faders and go from there.

Have you done mixes in the box that compare to what you can get from a console?

It's funny because when I first started I thought, "Man, mixing in the box sounds way better." Then I started listening to mixes that I really liked and realized that they were all done on an SSL. Then I started to mix like that and thought, "Wow, now my mixes are blowing away my in the box mixes. Just killing them." 5 years later I'd think, "Boy, that rough mix I did in the box sounded really good," so I've kinda gone back and forth. Then I started doing in the box mixes with analog outboard gear, then finally back to the console. At that point it was, "Why did I stop doing this?" It has the sound that I'm looking for so there's no reason to try anything else again. And I found that it's more fun pushing up a fader and turning a knob. It's way more fun that clicking a mouse or even something like a Faderport. That doesn't feel right to me.

How much processing do you do in the box as compared to the analog domain?

On a full session I probably have about 20 plugins internally. The only thing I really do in the box is surgical stuff. I do a lot of notch filtering or some sonic restoration or anything that I can do to correct any issues that the tracks might have when they come in. I use my own plugins on

everything, so there's a slew of those on busses going out to the console. Everything else is outside the DAW.

What are you using for effects then?

I have a rack with an SPX 90, a Lexicon 200 and a dbx 130 sub-synth that I use a lot. I also have a Roland Dimension D and a tape delay that I really like. Any kind of other special effects stuff I'll do in the box.

Do you have an approach to using EQ?

I do. I do corrective EQ first, like getting rid of overtones or annoying frequencies. Then I high pass and low pass so when it gets to the console I can use that for all the sonic sweeting to make things sound bigger and better.

So it's more boosting frequencies on the console and cutting in the box?

Yeah, you know when I first started mixing I learned to always cut first and get rid of stuff you don't need and leave the rest there. I did that for a while but I didn't like the way my mixes were coming out. Right around that time I had a single mixed by Andy Wallace. I went to the session and looked at the console and it was, "He's not afraid to add +15 at 200Hz on the bass guitar if it needs it." He had like +10 at 10k on the high hat. He got exactly the sound I was looking for. At that point I thought, "I'm doing this wrong."

Are there certain frequencies that you key on?

I know I like to have my kick drum as the lowest thing in the mix. I'm pushing 50 and 60Hz on that guy. I'm high-passing the bass guitar a little bit so it's accentuated at 80, while I roll off a little under that so it leaves room for the kick. On guitars probably around 120 to 220 or somewhere around there. I'll high-pass right after that to leave room for the bass.

Low end management is so important now. You have to find the areas where everything wants to be, but those are the areas that I like as far as the bottom. The top isn't as critical, I don't think.

Where do you come down on the air band? For some mixers it's really important and for others not so much.

Ted Jensen masters all of my stuff and he usually ads in a lot up there. I never really focus on the air bands up there. Maybe I should take a look at that [laughs]. The overall brightness is usually where I want it to be though.

Let's talk compression. I know you like it a lot. What's your approach?

I love it. Compression is my thing. I have over a hundred compressors in my control room, including the 52 on the console. You can never compress enough in my book.

When I'm tracking I love to have two or three compressors on many sources; something to catch some peaks, something to give it character, another maybe to give it some grit. At any given time I might use my Pawn Shop Comp (Dan's own plugin) on an insert in Cubase. I might also use the console channel compressor, and then on the insert of that

channel I might have my Altec limiter or a Gates SA-39 to give it some grit or some girth.

On my mix buss I'll have two compressors on that as well - sometimes three. My music buss is going to have an SSL compressor, and all of my busses will feed into the main buss on the console with maybe a pair of 1176's on the output. They're probably tapping 4 or 5dB with the gain reduction at 4:1. I can't get enough of it.

How do you approach panning?

I guess it depends on the instrument but overall I like wide panning. The drums will be panned to fill out how the overheads sound, so if in the overheads it sounds like a cymbal is over here, I'm going to pan it over here. If the floor tom is over there, then I'm going to pan it over there, so it reinforces it. Main guitars left and right with my widener on them. Vocal right in the center. If there's a doubled vocal then I pan it just slightly off to the left or right of the other. The same thing if there's a lead [guitar] or a harmony so they're not right on top of the lead vocal.

Reverb for the snare might not be super wide. Instead I'll pan in like 70-70. Same thing with a delay on a guitar or a vocal.

Do you replace drums often?

I do. Not necessarily replace but I try to supplement whatever is given to me with whatever is missing. I have a small arsenal of drum samples that have specific functions. If I get a kick drum in that has no bottom end, I'll throw in one that has a nice round bottom. If it has no attack,

I'll find one that has a nice click to it. Same thing with the snare. It's all about blending. If you replace things entirely you begin to lose the depth of the drums, if there was any to begin with.

Before you mix, do you have an effects setup that you always put up?

I try to have my console stuff set up as a template. I'll have my SPX 90 with a certain setting that I use for the snare. For vocal I'll have a couple of reverbs just for that - a small room to give the singer a kind of space to be in, and then a reverb to give it some tail or delay. Those are usually set and ready to go.

My dbx sub-synth is always on an aux for my bass track. The quicker that you can move, the better. After you do it for a while you know what it's going to sound like and you know what you want to do so you might as well have it set up.

How loud do you listen?

Not loud at all. I feel like you have to listen at a moderate volume so your speakers won't trick you. If you push them, you might not get the full frequency response from them and you might start making adjustments that might not be true.

Once I feel I have what's in the ballpark, I'll crank it up and do the hallway test and go out in the live room to hear how things are going. I never listen super loud sitting in front of the speakers though.

What monitors are you using?

I have a pair of audiophile speakers called Nola Boxers. I mixed on NS-10s forever like everyone else, and then I moved on to NS-4s (they sounded like NS-10s but with bottom end). I have a friend who does high-end home theater installs who turned me onto Nola speakers and it changed the way I heard things. I can't mix without these things.

How long does it take you to do a mix typically?

First song of an album usually two days, and then after that five or six hours to mix a song. If it's just a single then it's about eight hours.

How many versions of a mix do you deliver?

There's always a main pass, an instrumental, acapella, vocal up and down versions, and a TV track with the lead vocal muted. I might do a guitar up mix or a parallel drum buss up as well. There are about 10 or 12 different versions that I'll do, and then 20 or 30 stems in case someone wants to do a recall.

How long did it take you when you first started until you thought you were pretty good at it?

When I first started I spent hours and hours in the control room comparing my mixes to something I really enjoyed and going, "Why doesn't my mix sound like that?" That turned into months and years. Eventually you have great mixers like CLA (Chris Lord-Alge) or Andy Wallace mixing one of your songs where you go, "I know I can do this." It's that competitive spirit that made me want to get better and better at it. But to answer your question, it probably took about 10 years until I was confident in what I was doing. Hundreds and hundreds of mixes.

What would you suggest to someone who was just starting and just wants to get better?

I always tell people that there's no shortcut to proper engineering skills. You need to learn what sound is and you need to learn the craft. You have to learn what everything does and how everything works. Once you know that, do something a little bit different from everyone else. Don't try to copy someone else's style. Try to be you. For all the people that managed to stay in the game, you can pick out their mixes instantly.

How did you get started into plugin development, considering that you're a busy mixer and producer?

I feel like it was a natural progression. You want to move fast on your mixes and I wanted tools that helped me move faster. For all the processing that I do on my guitar buss, for instance, that all has to exist in one spot, and that brought about AIP (Amplified Instrument Processor by Korneff Audio).

This was all a long time in development. I thought about doing this back in 2008 or so and that's where I started to get into it by learning C++ and all of the programming basics. I put it on hold for a little bit, but then I started making sample libraries that required a little bit of programming. I though that if I was doing that, I might as well get back into developing a plugin.

Once I learned how programming works I started reading books about DSP and how all the signal processing worked. That took a couple years of my life [laughs].

I already had a pretty good background on the analog side. I love to design and build gear, so then it was all math from there to get to the plugin.

The Pawn Shop Comp was your first one. What was that modeled off of?

I wanted to do something that was different that didn't exist. My plan was for a tube signal path with an FET sidechain. I basically wanted tube grit with an 1176. That was the inspiration for that one.

Another inspiration was when I was first getting into recording I'd always go down to the pawn shop and see what they had. I'd pick up these DIY CB radio and broadcast boxes for like $20 that had knobs and meters that I had no idea even what they did. I was getting something unique because it was all DIY stuff and that was the idea behind the Pawn Shop Compressor - getting something that your not sure what it really does or how it works, but now you're tinkering with it.

The SSL Talkback compressor is very cool. You did a great job on that. What brought that about?

I had them in my console and use them all the time. When I first started getting into building DIY electronics that was the first circuit that I created. I used it on my drum buss for a long time.

I remember that SSL came out with their official plugin a long time ago but it never really sounded like the one in the console to me. I thought, "No one's got this yet, and even SSL hasn't gotten it right. Let's do it."

Of course after I released it I got a call from SSL, who were very nice. They said, “We’re glad you did it, but you really can’t mention SSL in any of your documentation.” [laughs]

It does give me some proud papa moments when you look through a thread online where people are comparing it to the SSL version and favoring it. It makes me happy that I did my job. I made something useful for me, and other people like it too.

What’s one thing that people don’t understand about plugin development?

I could name a thousand things but two things that people don’t realize is that there’s a real person with feelings behind it and when they blindly do things that are so hurtful online it really hits home. It’s actually rare to find something bad that people say about me or the plugins or the company and we try to keep it that way. We try to be nice to everybody all the time. That’s one aspect of it.

The other aspect of it is that everything behind a plugin is math. I never understood that until I started to read some DSP books. I would think, “These equations look a lot like what I was doing in calculus in school.” You read a little more and go, “This is calculus!”

Is there anything I haven’t asked you?

There’s one thing I want to add. Never take anything personally. If you do a test mix but the client doesn’t go with it or ends up going with another mixer, don’t take it personally. Most of the time, unless you mixes are terrible, those decisions have nothing to do with the sonics of what you’re doing. Maybe the decision is political in nature, or maybe

they just wanted to hear what you might do with it, or maybe it’s not that far away from what they were already doing. There are many things that are beyond your control.

Andrew Maury

Andrew Maury started his career in 2008 mixing front-of-house with Ra Ra Riot while producing and mixing small bands and artists between tours. He has since gone on to mix projects for Shawn Mendes, Post Malone, Lizzo, Kimbra, and many more.

Before you mix, can you hear the final product in your head?

I think on some songs yes, but I fully recognize that it's just the first thing that popped into my head. I expect it to morph and change as I continue and as other people weigh in, but I think that having some sort of vision for it is essential. How are you going to decide what to do otherwise?

You mix a lot of different types of music. Does your approach change for each?

Typically, no, actually. I think if I had to zoom in on my subconscious when I'm mixing, I'm really trying to let the production and song tell me what it wants and react to whatever seems to work.

What workstation are you using?

I came up on Logic, but I've been way more into Pro Tools these days. I think I mix about 75% Pro Tools now. Sometimes I'll still use Logic but only if it's something that I recorded there and want to keep it all in the same DAW. I like both programs though.

What's the reason why you changed over to Pro Tools?

The main reason was that I worked on some Sean Mendes stuff a few years ago. It was my first shot at doing a mix for a big pop artist and it was delivered to me in Pro Tools as a messy production session. I felt like I had to step up to the plate and finally learn Pro Tools. I came up in the late 2000s where Logic had just dropped its price down to $500 and Pro Tools wasn't the only game in town. I rode that wave as long as I could because it was affordable, but there came a point where I had to get into Pro Tools and I just threw myself into it.

Where do you start your mix from?

I don't have a formula with that. On a lot of the mixing I do, the rough mix is important, as it is with a lot of people. I think I start with the thing that strikes me as the weakest element, or the thing that needs the most help. Mixing these days is so much about picking up where someone else has left off. I think I've found more success with getting things finished and everyone happy by just running with where they're at and not by pulling the faders down and trying to explore a new balance.

If I have a really good personal connection with the artist and there's a lot of trust, I feel more inclined to take a more old school approach and really flex my intuition from the start. For a lot of new clients, if there's a session delivered with stems with a lot of processing built in, I take cues from them more than my own ideas first. That said, it never hurts to get the drums feeling good first [laughs].

Do you have to do a lot of editing and cleanup on sessions that you get?

Yeah. The thing I pay the most attention to is the lead vocal, of course. It astonishes me how little attention people put into crossfades between phrases and double breaths and all that. And there's quite a bit of de-clicking to do on basically everything. I understand the chaos that is working on a production in the DAW and moving quickly, but a lot of people miss a lot of bad edits.

Being a mixer I've developed a keen ear for pops and clicks and bad edits. When I work with artists as a producer where we're both looking at the screen together and we're building a track, they're always like, "How did you hear that?" I'm constantly chasing little edits and noises.

I always felt that it was part of production and should be done before you even see the tracks.

I agree. I think it's a mental bandwidth thing. It takes time to go back through all your work and check if everything is clean on each track.

Do you have a method for setting levels?

Sort of. I understand how elastic 24 or 32 bit is in the DAW. You can go in any direction up or down until the finish line. With that said, I always notice that I'm able to achieve a loud master when I have good headroom to work with on the master buss. I guess if I had to describe my instincts it's that I don't mind if a channel layer is hot or peaking up near zero, but as things progress into deeper busses I'm constantly bringing them down 6 or 8dB along the way. Ideally my master buss with no processing is clocking around -10 or -12dB FS. I find that the headroom issue is most important at the master buss and I think having a 10dB in a chain is what works best. The threshold of compressors react well at that spot and saturation kind of builds up nicely that way.

Do you use a lot of compression?

Yeah, but it's not prescriptive. Every sound is unique to itself and I'm really trying to hear whether compression will serve the song or not. I'll definitely use it, especially on the drum buss or individual vocals for sure. It's amazing how much compression you can get away with if you get the attack and release to play nice with the tempo and the performance. So I'm not afraid of compression at all but I recognize that it's a sound and not every song wants that.

Do you use it on the mix buss?

Yeah, usually. Some songs will only come alive the way I'm envisioning if I use a lot of compression at the mix buss. I tend to use a bunch of them, each doing a little bit. If it's not one of those songs that wants a lot of added energy at the finish line I'll just use something like the UAD Manley Vari-Mu and just lightly hit it.

The music will tell you what it likes and doesn't like, and I'm always learning more about that. Mixing in the box has really helped me realize that every song and every mix wants a different thing on the master buss.

Do you have an approach for using EQ?

Yeah. I use it for either problem solving or tone shaping. That's what everyone uses EQ for I guess [laughs]. I think of a mix as a puzzle. It's not only balance and panning and depth and effects, it's the tonal character that makes the puzzle pieces fit together. Sometimes I feel

that if just turning something up or down doesn't make it fall into place, you have to shape the tone in order to get it more audible.

I'm totally fearless with the EQ. The FabFilter Pro-Q3 makes it so easy to put it 25dB up somewhere and hear what that does. I remember when I first started mixing I was very careful with EQ, and now it's just like, "Whatever."

Are you one for wide panning or do you like to keep it closer to the middle?

I like wide panning. I also tend to play with an imager on the master buss to get a little extra psychoacoustic width so I'm always aware of how much that thing is doing and how much I'm building a mix with wide panning. I'm happy to go hard left and right as long as it feels balanced. I think if anything some of the people I work with ask me to narrow it in or make it not so extreme.

Are you working with live drums at all or is it mostly samples?

Yeah, quite a bit. I mix a lot of pop music that has no live drums but all my favorite stuff does. I come from a rock background so most times when I produce a project there's almost always live drums involved.

Do you ever replace drums or enhance them?

Yeah, I have Slate Trigger and I'm not afraid to use it. I think I learned the hard way many years ago when I first got Trigger how not to use it because it can be really distracting [laughs], but you develop an ear for it and learn how to make it best fit into the song. Having watched hundreds of interviews with mixers over the years, I can tell that no one

else is afraid of it either. I usually use it to add transients or frequency content that's missing. I'll try it without the sample first, but sometimes you just have to use it.

When you're producing, are you mixing as you go along?

Usually. Computers are powerful enough to juggle both these days. I like mixing the most so I have a tendency to start making things sound like a record right away. I know it's a balancing act because I've done productions where I've piled on the plugins and things just ran away from me and I wished I had a cleaner slate. It's a constant struggle. I'm working on a record right now where I'm battling it. It's like half of what I'm thinking about all day. I think, "Don't mix! Don't mix!" but then I just can't help myself [laughs].

Do you have an effects setup that you always use?

I do have a template Pro Tools session with a whole network of busses and effects. I'll import that into any mix job where I want to move quickly. My presets in that template tend to be a good starting place although I do tweak them per song.

If I'm working on a production from the ground up, I'll only add sends and busses if I really feel that I need to expand it in that direction. Sometimes I'll end up just committing that stuff into audio so I don't have to think about gain structure with it anymore. Producing and mixing both require a different headspace with effects and processing.

What effects do you generally use?

If we’re talking about reverbs and delays, it’s amazing to me how the Valhalla line has evolved and ticked almost every box. I love Valhalla delay. It also works almost like a reverb when you turn the diffusion up. It’s also low CPU so it hits the sweet spot for me. I could mix forever just with Valhalla, but I also use some UAD effects

With effects, are you going for depth or are you just trying to make everything sound bigger?

More often than not I’m using delay and reverb as a way to glue elements together. Along the way you achieve depth when you do that but usually I add it just because I feel something is too stark sounding. It’s disconnected from the rest of the track, and so reverb is a way to tie things together.

Do you have one reverb that you designated as “glue” that goes over a lot of the tracks?

Sometimes, but it can vary so much because if it’s a bright, fast, dense song, I’ll use a short room with a lot of brightness. If it’s a slow dreamy production it might be something like an 8 second plate with a huge low-pass filter on it. Using a darker reverb as the glue is something that I feel that a lot of amazing mixers are good at. It’s sort of felt. You’d never know they were there. It might even be the foundation of the song, especially in a slower more delicate song.

What monitors are you using?

I’ve got ATC 25s, Amphion one18s and NS10s. I also have a lot of ASC Tube Traps for the acoustics [Author: For the record, so do I, along with Amphion One18s]. I came up in Brooklyn doing this work and I’ve always had this loud voice in my head telling me, “Do not invest in a studio in New York City!” I’ve seen too many people lose their ass doing that, so I went with the Attack Wall [made up of Tube Traps]

early on, and I love it. It's come with me to four studios now and I've gotten nearly identical acoustic response for mixing. It really does feel familiar every time.

How loud do you mix?

I go all over the place. I love to rip my head off because it's fun, but I recognize that it can lead to some bad mix decisions and hearing loss. I usually work at about 80dB SPL for most of the day but sometimes I'll go to the point where my ATCs are bleeding [laughs]. I like to feel what I'm working on.

On the other hand, I'll also listen incredibly quiet. Every volume will tell you something different, and it's all valid.

I'm very aware of how our ears and brain compress sound. It's not quite the same as what a compressor does, but in effect it kind of is. I think the reason why we like the sound of compression is because it brings familiarity to the recorded sound. I want to know what the mix sounds like when my ears are compressing too because it's different. The frequency response changes. We're in the business of making something listenable for different people on different systems, so I like to explore it deeply.

How many versions of a mix do you do?

Some people sign off on mix one but that's rare. With some I have to go to version 15. I stopped nickel and diming my mental energy with that. One client might be picky and another might not be, so it will just average out in the end. I stressed myself out about, "This is taking too

long and there are too many revisions," or "I need to be getting paid more for this." I just found that thinking like that is distracting and disruptive to my peace of mind. Whatever anyone wants to do is fine with me now. I'm trying to get a reputation of being easy to work with. I think that's as valuable as getting a good sound.

How many mixes do you usually deliver?

If it's a major label I'll default to doing acapella, instrumental, TV mix and main mix. If it's an indie artist where I know they don't have some sort of deliverable requirement, I'll just make the main stereo mix and an instrumental. Everyone at least gets that, but major labels get more because they always ask for it.

Do you have favorite plugins that you find you use every mix?

Absolutely. I rarely pull up a plugin I haven't used before. As for compression on a vocal I love the UAD 1176 Revision E. I know that thing inside and out. I like the Waves Puigchild for a little bit of a squishing sound. I like the Waves API 2500 for more of a harder, clipped style of compressor. It's good on drums.

For EQs I like the FabFilter Pro-Q3 and I like the UAD Pultec. I can get most of my EQing done with just those two. For distortion I like Decapitator, Radiator, and Saturn 2. I'm a big fan of the Kush Audio plugins. They're super vibey.

How often do you use saturation and what do you use it on?

I'll play with saturation at any place in the mix chain. Ultimately I'll play with things and see what works. Drums is where it seems to be universally accepted. I like it on vocals but there are people in Pop music who really shy away from it. Bass can benefit massively from lots of distortion. Sometimes I do it parallel to the clean track.

I tend to feed all my tracks into four final busses - drums, bass, instruments and vocals. Sometimes in parallel to those I'll do one extra send which I call the "vibe" track. I'll feed all of those four into this thing and I'll use either Decapitator, or sometimes a compressor that's distorting, then I'll tuck it in underneath the entire master track. It's something that I can play with to add density and some fire.

I always want to achieve a mix that feels big and confident. I don't really care how I get there.

How long did it take you until you felt you were good at mixing?

The whole thing is a mirage [laughs]. I'm constantly wishing that I was achieving more than I have with my mix abilities. I think it was after seven or eight years that I was consistently getting work approved and genuinely liked the result I was getting - like wanting to hear my own work and feel proud of it. But I'm still in awe of the work that other mixers do. It's ridiculous to beat yourself up over not being a virtuoso at it every time you do it though. It's difficult.

What would you suggest to someone that's just started mixing in order to get better?

I really think that acoustics are important. It's not sexy like new plugins or powerful hardware, but I personally developed the most as a

mixer when I got my acoustics straightened out. I bought the Tube Traps and I began to understand what a really good stereo image sounds like and what a really tight low end response sounds like.

You don’t know it until you hear it, and most people haven’t heard it. Monitoring and acoustics are the foundation for being able to tell what you’re really doing.

Robert Orton

If you've enjoyed the big hits from Lady Gaga's The Fame and The Fame Monster albums, such as "Poker Face," "Paparazzi," and "Just Dance," then Robert Orton is your man. After spending eight years at the side of producer extraordinaire Trevor Horn, Robert has gone on to craft hits for Robbie Williams, Enrique Iglesias, Carly Rae Jepsen, Flo Rida, Kelis, Usher, Mary J. Blige, and Marilyn Manson, among many others, while winning a few Grammy awards for his work along the way. Robert is also one of the first hitmakers influenced by earlier versions of this book. Says Robert, "The Mixing Engineer's Handbook is one that I remember studying very closely when I was an aspiring mixer hoping to catch a break."

Can you hear the finished mix in your head before you begin?

I think that having a pretty clear vision of where you want to go from the beginning is one of the most important things in achieving a good mix. Having said that, I don't always work that way. Sometimes I just see where the mix takes me, but a lot of the time I'll immediately think, "I know what this needs," and have a fairly clear idea of how I want it to sound. It's all about being bold and making big decisions.

Are you mixing mostly in the box or on a console?

I'm mostly in the box in Pro Tools with a load of plugins. I've got a few bits of outboard gear as well, mainly a bit of compression, like some [Empirical Labs EL8] Distressors, because there's not much that sounds like them. Sometimes you just need a bit of analog to find the sound you're looking for, but for the most part I'm in the box. I guess

I've always really mixed that way. I like the way it sounds, and it's so much more convenient when it comes to recalls.

Are you using an analog mix buss?

No, I've tried those kinds of things, and I haven't felt that I've gained a lot from using them. Basically I mix in the box, and then I've got a couple of things that might go over the mix, but I don't buss it out or stem it out or anything like that.

Do you use a controller?

No, I just use a trackball. The few times when I mixed on an [Avid] Icon or that sort of thing in the past, I've found that I ignored it most of the time. I'm so used to doing rides on the trackball that I forget that the controller's there. I have a really simple little setup. I just sit in front of a pair of monitors with a computer in a well-treated room.

Where do you start your mix from?

I start with all the faders up. The first thing I do is listen to the rough mix of the song I've been sent a few times to get the idea of what the producer and artist intended. Then I just push all the faders up. I like to mix while hearing everything. Once I've got a balance that feels about right, then I might listen to the drums and see how they're fitting with the bass and see how the vocals are sitting on top of that, but I wouldn't say that I build a mix up by adding one instrument at a time.

Do you augment the drums, or is what you're given what you work with?

I try to work with what I'm given just because augmenting can change the character unintentionally if you're not careful. Having said that, sometimes you get a kick and think, "This is never going to sound how I want it," so you have to add something in. Of course, that happens more with something like a rock band.

There always seems to be a lot of layers in your mixes. I know you started with Trevor Horn, and his stuff is like that as well.

Is that where you got that technique from?

I worked with Trevor for eight years, and I learned so much. He's so talented and just an unbelievable guy. I learned something from him every day I worked for him, and I'm sure that if I worked with him again, I'd still learn something new from him. It's true that he layers a lot of things. Most of his multi-tracks tended to have a lot of parts, and mixing that stuff was difficult in terms of the track density, so I got used to layering things and figuring out how they fit together. That rubbed off on me, and I certainly do approach mixes like that now.

You're not afraid to go wet on a mix even though the trend is to stay dry.

I just do what feels right, really. It's about creating shifts in perspective and excitement between different sections of the song. There's definitely a lot of that going on in the Gaga stuff, with extra delays that come in on the chorus and that kind of thing. Dry is good because it can be in your face, but a bit of wetness can position things and stop them from bumping into one another.

You seem to use a lot more delay than reverb.

Yeah, I don't use much reverb at all, actually. Sometimes I'll use it more as an effect if there's a big moment that needs some sort of emphasis, but generally I try to stick with delays because you're left with more space in the track.

Do you use a set number of delays right from the beginning of the mix?

I do have a template that I start from, but to be honest, everything's individually tailored for each mix. I don't really know what I'm going to need until I start diving in. I normally just listen to it and tweak as it's needed. I spend a lot of time on the vocal because that's often the most important element of a mix. I add effects while listening to the vocal in the track, because what's on a vocal can really influence the way the backing tracks feel against it.

I notice that you use a lot of stereo delays as well.

Yeah, I do that to create excitement in the mix. You've got frequency to use and you've got depth to use, but certainly there's width that you can create excitement with, too. Things that sit far left and right in the mix can make it sound more exciting. Having different delays on either side can often make a vocal sound bigger, but at the same time they don't get in the way of the direct sound.

Do you have any plugins that you almost always use?

Yeah, I do. It's funny; you do tend to go toward the same things that you like the sound of. I love all of the Waves plugins, especially the modeling ones that they've been doing lately. The Chris Lord Alge compressors sound great, and I use the Neve and the API EQs a lot. I use [SoundToys] EchoBoy a lot for delays, and the Bomb Factory Moogerfooger is really useful.

I try to push myself to use the plugins in creative ways instead of the obvious ways that they're always used. For problem-solving I tend to go toward plugins that I like the sound of and I know what they're going to do, but sometimes you're looking for a vibe when something's not happening in the parts that you're given. That's when I'll look for something more creative with a plugin, like an amp model on a vocal or an automated sweeping filter over the backing track. The sort of thing that completely changes the sound and creates something new for you to feature or lock onto in the mix.

Do you have a particular approach to using EQ?

I use a number of approaches, actually. Sometimes I'll hear a frequency that I don't like; then I'll use some subtraction. Other times I might think, "This just needs to be brighter," and I'll just turn the knobs until it sounds right. I don't think too much about the frequencies, and I'm definitely not afraid to boost a lot.

Having said that, I don't just go in and EQ everything. I think the best advice I could give anyone who wants to mix is to learn when not to process something. The more you process things, the worse it sounds, I generally find. To me, mixing is more about balance and groove and getting that to happen. I always try to get the mix to sound as good as I

can just with the balance before I go in and start EQing. I might fix something at the beginning of a mix if it feels wrong to me, but I don't just jump right in and EQ things. I think that leads to more of a mess than anything.

How much compression do you use?

I'm quite light with it. I do often use some compression over the mix buss, but I don't automatically put it on. I find that sometimes compression helps the groove of the song. I don't really use compression to hold things down in their place unless there's an obvious problem. I might compress a lead vocal quite heavily to maybe bring out a certain energy in it, but generally I don't compress a lot. A lot of mixing for me is about feeling the movement and the groove, and a little bit of compression in the right area can help certain elements fit in the pocket and make that happen, so I might use a little for that.

The Gaga stuff is a really good example. "Just Dance" has a slightly odd drum pattern that sounds a bit like four on the floor, but the four was actually just the snare. I used an SSL buss compressor across the mix, and it really helped that feel because of the way the kick and the snare reacted to it. That's a good example of how buss compression can influence the groove of a song. I use compression more like that, really. I don't automatically reach for it, but sometimes it can really help accentuate the groove.

What's your approach when you have a song that uses a synth instead of a bass guitar for the bottom end?

I think you have to approach it slightly differently, because quite often those types of synth sounds interact with other parts of the song. Maybe the top end of it is buzzy or something like that, and it's not

always just the traditional low end that you're looking for from some of those parts. Sometimes I approach that by maybe multing it out so I've got the bottom end from it [on one channel], but treat it in parallel and then blend the sounds together.

I've noticed that in those kinds of songs you've made the kick a little bigger than it might normally be, so it takes up a lot of those frequencies as well.

I tend to mix the kick drum quite loud in that style of music. I know that's a bit of a trend at the moment as well, but I like to hear it thump you in the chest. Sometimes I'll use a trick like compressing the bass every time the kick hits so you don't need to balance the kick as loud for it to have impact.

So you're keying it, then?

Sometimes. That's certainly a good trick when you have a dense bass synth that's taking up a lot of space. Sometimes that kind of trick can open up space in the mix if done subtly enough.

How long does it take you to do a mix?

I do a mix in a day. I normally start at noon and keep working until it's done. Sometimes I go home at 10 in the evening, and sometimes I go home at four in the morning; it depends. When I'm not under a really strict deadline, I like to come in and listen the next day as well. Going away and getting a fresh perspective with a new pair of ears can be very valuable.

What monitors are you using?

I'm using some ProAc Studio 100s, which is what I listen to most of the time. I also have some Quested VS2108 midfield speakers. I also love listening on a laptop just to hear how it sounds on something really small, which is the kind of thing that people might listen to music on a lot these days.

Do you listen on earbuds?

No, I don't. I do have a pair of Sennheiser headphones that I really like. I use them right at the end of a mix just to be sure that I haven't missed any clicks or anything like that. Sometimes things jump out at you a little more obviously on headphones.

How loud do you mix?

I monitor really quietly most of the time. When I'm getting the drum sound and again toward the end of the mix,

I might give it a little blast on the bigs to just kind of check the bottom end. I might turn it up when I'm doing the vocal to see if there are any frequencies that catch my ear, because sometimes when you turn it up the sibilance will irritate your ear a bit more.

There's only a few times during the mix when I turn it up, though. The rest of the time I monitor really quietly. If there's someone else in the room tapping on their computer, it bugs my ears. [Laughs]

How many versions of a mix do you do?

Unless someone specifically asks for a vocal up or vocal down, I normally just print the mix, an instrumental, a TV mix, an a cappella, and some stems. I kind of prefer it when the mix I give them is the definitive version and there's no question about it.

How much do you automate things?

I do quite a lot of automation. I spend the first part of the mix getting a rough balance, but I spend the second half of the mix just riding everything. Balance is certainly a dynamic thing. I might change the feedback on a delay or alter the filters on a section or ride the effects sends, and I certainly do a lot of volume rides.

Are you drawing it in or using the trackball?

Mostly with a trackball. I do draw in some things early in the mix, because maybe one sound might be too loud or too quiet in a certain section, so I do broad brushstrokes at that point. As the mix is coming together, I'm doing a load of rides with the trackball just by feel.

I mix in quite an odd way, I think. I sort of listen to each section of a song repeatedly. I might listen to how the pre- chorus goes into the chorus, and I'll go over that one section quite a lot of times and push different things up, riding them with the trackball until it feels musically right; then I'll go to the next section. I'll sort of work my way through the song like that rather than doing full passes from beginning to end. One of the reasons is I'll think, "I've got to go back and fix that

thing I just heard." If I let it run to the end of the song, I'll have forgotten about it. It's a good way to focus myself.

What have you learned between the time you just started to mix full time and now?

I don't think I can pick one thing. I think the amazing thing about mixing and the thing I love the most about music in general is that every day I learn something. I'm always surprised by how sounds interact with one another, and if you change one thing it might change something that you didn't expect. The way that music works and the way we perceive it is a constant amazement to me. I feel like I learn something new every single day.

Greg Penny

Born into a music business family to bandleader/producer Hank Penny and hit recording artist Sue Thompson, Surround Music Award winner Greg Penny seemed destined for a life in the studio. Indeed Greg's production aspirations resulted in hits with k.d. lang, Cher, and Paul Young, among others, but a meeting with Elton John while Greg was in his teens turned into an award-winning mixing journey with the legend many years down the road. Greg gives us an inside look at his long relationship with Sir Elton as well as some insight into his sensational surround remixes. You can find out more about Greg at flowerrecords.com.

How did you get connected with Elton?

I met Elton backstage at his first Las Vegas gig, and then the second time he came through I brought my Mom down to see him. He was a fan of hers, so he was very happy to see us. After that I stayed in pretty close contact with him, and I'd go to his gigs when he'd come through town, so he knew that I was a real sincere fan and that I wanted to be in the record-making business.

When I was 17, I decided to go to Europe, and Elton invited me to stop by the Chateau in France to check out what they were recording, which was Goodbye Yellow Brick Road. The day I got there, Dee [Murray—Elton's longtime bass player] was fixing the bass track to "Saturday Night's Alright for Fighting." They cut the track the night before, and he was just patching some things up.

That's an awesome education right there.

It also had a lot to do with me inheriting Gus Dudgeon's [Elton's producer] projects because I had so much respect for him and he was one of my heroes. Gus let me sit in on stuff and never shooed me away. He never had an attitude with me and was always encouraging because he knew that this was what I wanted to do, so he gave me the space to hang out and be a fly on the wall. I didn't go to London to the mix sessions for the record, but I stayed pretty up to the minute on everything they were doing.

Over time we've remained friends; then in the early '90s I produced these records with k.d. lang that were real successful, and it turned out that Elton was a fan of them. He rang me one day and said, "I want to do this album of duets. Would you do a track with me and k.d.?" At the tracking session he asked me if I'd do more songs on the duets album, which I did, and then he asked me to do his next studio album, which was the Made in England album that became really successful.

That kind of got me to where all I did was Elton stuff for about three years, from studio to live stuff to remixing other things to following up on B sides and things for charity albums. There was just a lot of work, and I gladly fell into it and kept on going. A couple of years ago I went back to him with the idea of mixing his catalog in surround, and he was really open to the idea.

Do you incorporate the mixing tricks you learned from Gus now on your more recent stuff?

Yeah, more often than not I do because I find that when I listen to mixes where I've been subtle or timid about the level of a part, I tend to later feel that more dramatic colors or more dramatic mix rides had to

be done. I've gotten more radical with things, and when it's time for them to be featured, I'll crank 'em up.

Let's talk a little about your general mixing philosophy.

I usually start a mix from the vocal. Obviously there are instrumental things that you have to be concerned with, but my thought is how the vocal is going to work with the elements that are supporting them, so I guess I mix from the vocal out.

Sometimes I'll poke around on drums and bass to get a good floor and then I'll take them out and put the vocal in to try to get a really good sound on it; then put all that together with all the other elements around it, always bearing in mind that the vocal has to be the thing that pulls people in. For me, that's really the most important thing, although that's not necessarily true for everyone who mixes or all artists. There are grooves from each artist that are special to them, but they're nothing without their vocals.

It's been fun to do the surround thing because I've been able to highlight things just by virtue of the fact that in stereo things get crowded. In surround I've been able to pull out instrument passages around the vocals.

What's your approach to mixing surround?

It really started with Goodbye Yellow Brick Road because I had a few weeks to sit with it and come up with a formula. I thought, "Here's this

incredible record that's got tons of guitar and keyboards on it, so why don't I put the vocal in the center monitor most of the time, and the only other things that enter into that monitor are double vocals or harmonies or maybe even a solo instrument? Then I'll bleed out a little bit of the center vocal into the left and right fronts, so if Uncle Bob comes over to the house and sits at that end of the couch he's not missing the lead vocal. Then I'll use divergence and spill a little of that lead vocal into the rear monitors also for that purpose. Or we could give Elton the front of the room for his piano and Davey the rear of the room for his guitars, or even put them in quad like in 'Saturday Night's Alright for Fighting'—four guitars, four speakers, with each guitar getting its own speaker." That started me moving in that direction, and then it got more radical as I went through the album. By the time I got to Madman Across the Water with those incredible string arrangements by Paul Buckmaster, it was amazing to put the whole string section in the back and the band in the front. They recorded everything at the same time, so when you listen to it it's like you're sitting midway between the band and the string players, just like on the original date.

It sounds like the fact that everything was cut live greatly influenced your surround mixing philosophy.

As Nate Kunkel says, you've got to make records for the guys who have the right system, because if you back off from that and try to dilute the vision down to the most common denominator, we'll never get anywhere. You have to somehow set the benchmark. More often than not on these blogs on the Internet about Elton's mixes, I either get slammed for being sacrilegious or lauded as a genius. They always talk about how the use of the surrounds is extreme on Elton's records, but then a lot of guys say, "Finally, I'm able to hear my system because I've got some stuff that puts something through all of my speakers." My objective is to use the system entirely and not to be too timid, but there

are some songs where you just don’t have enough data to put in all the speakers and have it make sense without it seeming lopsided.

What kind of gear are you using?

I use a Pro Tools HD system. All of Elton’s stuff is at 24/96. It seemed that the jump up to 96k was much more noticeable than up to 192k. I have a very simple single interface with a [TC Electronic] System 6000. I mix in the box. On occasion I’ll go out to a little TL Audio tube desk to try to get analog warmth out of some things if I can’t get it within the box.

For speakers I use Dynaudio AIR6’s. I come right out of the back of the HD interface right into the back of the speakers with nothing in the way because the controller that comes with it is sufficient for me. I’m not doing any downmixes with the mixes that I’m doing, and if I need that I can test it with a Waves plugin across the buss rather than having something like a StudioComm box [surround monitor controller]. The Dynaudios are incredibly accurate right down to a whisper level and then can also get incredibly loud. The system also translates very well when my mixes are played back in other rooms.

Are there any tricks you’ve found to make mixing in the box sound better for you?

I use a pretty simple set of tools inside. I don’t use a lot of on-board plugin reverbs. I have a couple of short things that I use, but predominantly I use the System 6000 surround reverbs. If I don’t have enough firepower with the one unit, then I’ll print the reverbs and go back and do it again.

I use the Waves 360 suite a lot. I use their panner because I find it's easier to automate when I can just grab the toggle thing and fly it around. I also like the L360 limiter.

It's actually a very simple set of tools that I use. I don't believe for the albums I'm doing that I need to get complicated, because when these albums were originally mixed they only had a couple of plates available, maybe one that was a second or so long and then another one maybe about two and a half seconds. Over time they got a digital delay, and that's how they did the handclaps on "Benny and the Jets," but most of it was tape slap. They used very, very basic things, so I try to stay with very basic stuff. If I do use anything high-tech, I try to make it sound transparent.

Has the Elton stuff influenced how you work on other things?

Yeah, it has. It's influenced me to stay simpler because I think I can get better imagery. I remember hearing Elliot Scheiner saying at a surround conference, "More often than not, if I want something to be in a speaker, I put it there. I don't often try to occupy the space between monitors [phantom image], because I find I get a lot of phasing and strange clarity issues." I've adopted my own version of that. It depends on the instruments that I'm using, but I have a fairly formalized way of placing things now, and it seems to work. I know where I like to put drums, I know where I like to put the bass guitar, all the way down to background vocals and strings and things like that, although I never start a song with a prepared template. I always start a song from the ground up, and depending on what the song dictates, I'll change the placement.

Do you think you'll still stay in the box for all your projects?

I would like to do some things that are initiated in surround and are repurposed later into stereo, which is sort of the reverse of the way things are done. I'll probably stay in the box for that not only for simplicity, but because that's where the project started, and it will probably organically take shape that way.

I think it would be different if I had a different budget structure to work within, but there just isn't a lot of support financially for mixing in surround, so you have to find a way to be clever if you want to do it, and that makes me stay in the box. Oftentimes now I'll work on something for a half an hour, and if it's not happening I can just pull another track up, so I can work on three or four songs a day. Nothing ever gets boring and nothing ever suffers because you're trying too hard, which I think is really the beauty of it.

Dave Pensado

Of all the genres of music, mixing R&B may be the toughest, thanks to the almost constant change in what's state of the art and the artists' and producers' penchant to experiment with new sounds. Over the last two decades, Dave Pensado has taken mixing to a new level in artistry, having mixed big hits for superstars such as Christina Aguilera, Justin Timberlake, Kelly Clarkson, Pink, Black Eyed Peas, Beyonce, Shakira, and Michael Jackson, among many others. Well known in the business way before his popular Pensado's Place web series, Dave not only is on the cutting edge of technology, but also has thought long and hard about the more cerebral aspects of mixing. For more about Dave, check out pensadosplace.tv.

What's harder to mix—an R&B or a rock track?

I mix both, and R&B is infinitely harder to mix than rock. Think of it this way: Let's say you're painting a portrait. Rock is like having the person you're painting sitting in front of you, where you look at them and paint, and look at them and paint, so you always have a reference. In R&B, there is no reference. It's like trying to do a portrait from memory, but because you don't have the person there, you can paint something that transcends what he is. You can make him prettier, you can make him uglier, or you can make him abstract if you want.

R&B gives you fewer limitations and a lot more freedom. We don't have to have the snare drum sound a particular way. It can sound like anything from an 808 to a hand clap to a little spitty sound to a rock sound; but you put certain snare sounds in a rock song, and it's just not a rock song anymore.

Do you approach EDM differently, too?

Yes, I do. I would like to think that I approach every mix differently, much less every genre. When I'm changing styles of music, I tend to immerse myself in that type of music for a day beforehand so I can clear my musical palate, so I'll listen to a lot of EDM just to get in that frame of mind. The other thing is that when I'm starting an EDM mix, I like to make sure that I have the newest, hippest plugins around. In that genre sounds move so fast that you have to have the freshest stuff to create separation between you and the latest songs that are out there.

Do you hear the finished product in your head before you start a mix?

Yeah, I really can. I might not have 100 percent of the final product in my mind when I start, but I pretty much have it outlined. Then as I start filling in the outline, sometimes things change a little bit. Every once in awhile, maybe one out of two or three hundred, I might just pull everything down and say, "I don't like any of this," and start again from scratch.

What's your approach to using EQ?

Well, I think of EQ as an effect, much the same way you would add chorus or reverb to a particular instrument or vocal. I tend to be most effective when I do the standard equalizing; then take it to the next level. For instance, I might take a vocal to where I think it's really EQ'd nicely, but then I might add a little more 3k just to get it to bite a bit

more so it brings out a little bit of passion in his or her voice, just to make me feel like the singer was trying harder.

Are there certain frequencies that you keep on coming back to?

I notice that in a broad sense there are. In other words, I always have to add the frequencies from, say, 120Hz down to 20 cycles. It seems like I'm always having to add the frequencies from about 10k on up as well.

So much of the music that I do is programmed, and that gives the producer the luxury of pretty much getting the sound he wants from the start. In the old days you always pulled a little 400 out of the kick drum, and you always added a little 3k and 6k to the toms. That just doesn't happen as much anymore because when I get the file, even with live bands, the producer has already programmed or triggered the sound he wanted off the live performance, and the drums are a lot closer to being finished. It frees me up because now I have the luxury to really get inside the tracks within the timeframe I'm given, whereas before I would have to spend that time just getting it up to a certain level. Most of the stuff I'm given now is really starting out in a lot better shape than it used to sound-wise.

How about panning?

I think that there are three sacred territories in a mix that if you put something there, you've got to have an incredibly good reason—that's extreme left, center, and extreme right.

I've noticed that some mixers will get stereo tracks from synthesizers and effects, and they just instinctively pan them hard left and hard

right. What they end up with is these big train wrecks out on the ends of the stereo spectrum; then they pan their kick, snare, bass, and vocals to the center and there's all this stuff stacked on top of each other. If it were a visual, you wouldn't be able to see the things behind the things in front.

What I do is take a stereo synthesizer track and just toss one side because I don't need it. I'll create my own stereo by either adding a delay or a chorus or a pre-delayed reverb or something like that to give it a stereo image. I'll pan maybe the dry signal to 10:00 and then I'll pan the effects just inside the extreme left side. I would never put it hard left because then there are too many things on top of it. I might pan it at 9:00 and then pan the dry signal to, say, 10:30 or something like that.

Do you use a lot of compression?

I look at compression as having two functions—as an effect, and when you want to keep a particular sound right up front in your face in the mix. I very rarely use a compressor to even out dynamics. Dynamics are something that I just can't get enough of. I once read an interview with a well-known engineer in which he was praising a particular compressor for its ability to take the dynamics out of a drum performance because the drummer would get happy on the first downbeat of every chorus and play a little louder. I thought, "I spent my whole career trying to add those dynamics and trying to make the drummer sound like he got happy going into the chorus."

The compressors I like the most tend to be the ones that actually help me get some of those dynamics. That might be a contradictory statement, but if you're careful with the attack and release times, you can actually get a compressor to help you with it.

How do you do that?

A lot of times what I'll do is put the effects only on the compressed sound. As a result, the reverb actually has a snap and aggressiveness to it. Every once in a while I'll make it stereo, where I'll put anywhere from a 9- to 15-millisecond delay on one of the channels so the tight compressed sound is out on the edges of my stereo spectrum, but the original sound's in the center. That creates an incredibly nice image, particularly for ballads and slow tunes where you have a lot of space between the downbeats. That setup works great for snares, kicks, and hi-hat. Every once in a while, it'll make a guitar come alive, too.

What you're doing is controlling the dynamics, but you're actually increasing the dynamics, too. It's the strangest thing because psycho-acoustically it's not getting louder, but your mind is thinking it is. On the radio, it just jumps out of the speakers.

Do you use much mix buss compression?

I rarely do anything to add color on the master buss. I was taught that putting things on the master buss is an excuse for not doing something earlier in the chain. If you have to add a lot of anything to the master buss, then you probably haven't worked enough on the individual elements. For me, a dB here or there is okay, but I try to keep things minimal. If I'm in the box, I'll try to use some [Waves] L2, but if I'm not in a box, I'll use the SSL board compressor if I use any at all.

How much of your mixing is in the box these days?

About 60 percent. It's all dictated by the budget. Sometimes even the clients that can afford it would rather I mix in the box to save some money. I think I can get it pretty close to the sound of a console—maybe to within 2 percent or so.

Do you have a philosophy about adding effects?

The way I think of it is the pan knob places you left to right while the effects tend to place you front to rear. That's a general statement, but it's a good starting point. In other words, if you want the singer to sound like she's standing behind the snare drum, leave the snare drum dry and wet down the singer, and it'll sound like the singer is standing that far behind the snare drum. If you want the singer in front of the snare drum, leave him dry and wet down the snare drum.

That said, I like a vocal mostly dry, but then it usually doesn't sound big enough. You want the vocalist to sound like they're really powerful and dynamic and just giving it everything, so I'll put an eighth-note delay on the vocal but subtract a 16th-, a 32nd-, or a 64th-note value from that eighth note. What it does is gives a movement to the delay and makes the singer have an urgency that's kind of neat. I put the eighth minus 1/64th on the left side, and put the straight eighth note on the right side. You can experiment with pushing the pitch up a little bit on one side and down on another, too, if your singer's a little pitchy, since that usually makes them sound a bit more in tune.

Sometimes putting the eighth-note triplet on one side and the straight eighth note on the other, if you've got any kind of swing elements on the track, will make the vocal big, yet it doesn't make the singer sound like he's taking a step back.

Another thing I like to do is to take the output of my effects and run them straight into another effect. I'll take an exciter and just dump the output straight to a chorus so it's only chorusing the high frequencies. I think that's more pleasing than having low notes chorusing all over the place. Another thing I'll do is set up a chorus, and I'll pan one side hard left and the right return at 2:00, then I'll take another chorus and pan it hard right, and then the left return from that one I'll pan at 10:00. Now the lefts and rights are kind of overlapping. On one I'll have the chorus depth just a little less than the other, and I'll have the other modulating a third faster. When you add a vocal to that, you get this real nice spectrum that just widens it out because there's an equal amount of chorus, yet one of them is chorusing deeper and slower than the other one. If that's not wide enough for you, add a delay in front of both of them that's different on each side and then add that to your background vocals. They don't take any steps back in the mix, but they just get fat.

A lot of times I'll take two reverbs and instead of running them stereo, I'll run them mono in and mono out and pan one just inside the left and one just inside the right. I'll use the same program on both, but I'll slightly alter the values. Just return it in mono, and you'll be surprised by how much better it sounds sometimes.

What operations do you still do in the box even though you may be mixing on a console?

When I mix on a console, I mix 100 percent differently from when I mix in the box. When you work in the box you have to make some decisions about how you're going to make it sound, especially if you have someone who's used to listening to analog, because you don't have a console to supply that sound. You have to give the producer or artist something that they've been accustomed to for a long time.

Immediately you start thinking in terms of the tape machine emulation plugins (my favorites being the Waves and UAD, although Steven Slate just came out with one that's good). I might take my drums and put them on an aux, then put a tape machine plugin across that aux. Or I might use a plugin by Dave Hill called Phoenix to add some color or use the AC series from McDSP. You have to practice with every one of those saturation plugins to see what they do best because there's not one flavor that will work in every situation.

Are there any go-to plugins that you always use?

There are probably too many to list. [Laughs] I use the P series and sometimes the E6 from McDSP. I use the Waves plugins that everyone knows: the C1 series, the Renaissance Compressor, and TrueVerb. I use the Massey L2207 on just about every mix. I love his TD5 Tape Delay as well. I also use the stock Avid plugins. Those are monsters, and they're very efficient. I'm a big fan of the Air series.

At some point in the mix you have to start choosing plugins based on their efficiency instead of only their sound. There are some that take up so much processing that you can't use them if you're at the end of the mix because you'll run out of processing power. Anything from UAD, Steven Massey, or McDSP works well in that regard because they're all so efficient.

How long does it take you to do a mix these days?

If I had to put a number on it, I'd say that hip-hop takes about 10 hours; pop about 14 hours; rock that's recorded well about 10 hours—and if it's not recorded well, then maybe 16 hours; and EDM anywhere from 12 to 18 hours depending upon the subgenre.

The amount of time is directly proportional to the quality and the number of tracks that I'm given. If doing a mix with four different lead singers, that's going to take me a lot longer than doing one with a single rapper or an instrumental EDM song. Another thing that slows me down is when the session doesn't sound like the producer's vision. In other words, when I pull the session up it sounds like one thing, but when the producer comes in he has a completely different idea of how it's supposed to sound. Those take a long time and add maybe about 50 percent more time to the mix.

What's the average number of tracks you're given?

I never really count the tracks, but when I first pull up a session I have to make a decision on how many voices I want the DAW to use. I'd say it's rarely ever fewer than 90 and usually averages between 120 and 160. The song I'm mixing now I'm struggling with because I only have 192 voices on my rig, and that's not enough!

Is this because they expect you to figure out what to use from all the tracks?

In the world that I work in, one of the major changes over the years has been the removal of any need for commitment. Back when I was working for Babyface, he would record six to eight reels of background vocals [which might have up to 20 tracks of vocals on each reel], which is a lot of tracks, but I never saw them because he bounced them all down to two or three stereo tracks for me to mix. Nowadays I never see anything like that, and I haven't seen it for at least five years. I get every single track that's been individually recorded. Nobody wants to commit anymore. That's okay, I don't mind doing it. It gives me a little more flexibility, and we usually end up within

5 percent of what the producer wanted anyway.

Another change from the old days is that the rough mixes should probably be called "reference mixes" because there's so much time spent on them. Sometimes they'll spend two or three weeks on one and even bring in an engineer specifically for that process. They're not rough mixes at all; they're pretty much what the client wants. I think it has made us all better mixers, because 10 years ago the gap between what they gave me and what they got back was a lot wider. People would walk into my control room and say, "Holy crap! That's incredible!" You don't get as many of those moments anymore, because they've spent so much time getting it so close before I even get it. The good news is that allows me to spend more time on the small things to make them a lot better.

The other big change is that five years ago, if you didn't like something, the tiniest change would end up costing at least five grand, which isn't the case now because you never really finish a mix that's in a DAW. I get calls on mixes that I completed six months ago, asking me to change a word in a chorus. I don't think that's a bad thing, but it all goes back to the lack of commitment. I think creativity works best when there's a deadline. We're never in a finished state now, because everyone has a DAW at home and knows how quick it is to pull something up and change it.

What alternatives mixes do you deliver?

Ninety-five percent of the time they just want the main mix. If they want the vocal up, they just call, and 10 minutes later they have a mix with the vocal up. In terms of the classic stems of a cappella, TV mix, and instrumental, I don't even get many requests for that anymore unless the artist is performing live. Sometimes they ask me to send the entire session over to the live mixer, which I don't understand because I'm using plugins that not many of those guys have.

What monitors are you using these days?

I hate to bore you, but it's still NS10s and Auratones. If I'm working at Larrabee [Dave's Hollywood-based commercial studio of choice], I've got the Augsburgers [main monitors] and all the rental monitors available if someone wants to hear it like that. [Producer] Ron Fair gave me a pair of KRKs with a KRK sub so he would have that comfort factor when we work together. I encourage that from all the producers I work with. [Laughs]

The NS10s give me all the low end I need. When you see the specs on a speaker like an NS10 that says it's down 20dB at 80Hz, that doesn't mean that you can't EQ 80Hz, because it's really there; you just have to train your ear to hear it at 20dB down and not flat. To get your ear trained just takes a while until you do some mixes and hear what they sound like in a number of places, then determine what the strengths and weakness of that speaker are, and then mix accordingly.

Do you have any listening tricks?

Yeah. I have my sub connected so I can monitor it without hearing the mains to know where my low end stands. I've done it so much that I can get the low end right just by listening to it that way. Then sometimes I walk into the hallway to listen to a mix, and that helps me get some balances, and the old car test has always been a great tool that we all use. Now more than ever, it's integral to the process if I'm working in someone's private studio.

What level do you usually listen at?

I usually listen to NS10s at kind of a medium level, and Auratones I listen to at the same volume you would listen on a TV. I found that in order for the NS10s to really work, it's best to have them stay at one level for most of the mix. Near the end of the mix, I'll check the EQ with the NS10s about 20 percent lower and again about 20 percent higher.

When I'm at Larrabee, I'll use the big speakers mostly to show off to clients and to just have fun. I like to turn it up, and if my body is vibrating properly, then I'm happy with the low end. A lot of engineers use them to hype the client, but I also use them to hype myself! If I'm cranking and I'm not getting excited, then I just keep on working.

What I recommend is to train yourself to work at three volumes and mark them so you work consistently at them. After a while you'll do it without even thinking. That's what you want from the entire mix process. You want to be removed from any technical thoughts and be 100 percent in creative mode.

What advice do you have for someone who wants to get better at mixing?

Keep an open mind to all music. You'd be surprised by what you can learn from records outside of the genre that you normally work in. Focus on things that can expand your life, and they will expand your creativity as well. Life experiences are always going to contribute to your creative process.

When it comes down to it, emotion and passion are at least as important as technique. The way I can prove that's accurate is the number of times a rough mix gets used over the mix of a top-level

engineer and goes on to sell millions of records. Creativity is about the spur of the moment and not thinking about the process. Emotion always trumps technique and technology.

Elliot Scheiner

With his work having achieved tremendous commercial success, Elliot Scheiner has also attained something far more elusive in the music business—the unanimous respect of his peers. Indeed, if you want a mix that's not only a work of art, but also a piece of soul that exactly translates the artist's intentions, then Elliot's your man. With a

shelf full of industry awards (seven Grammys, an Emmy, four Surround Music awards, the Surround Pioneer and Tech Awards Hall of Fame, and too many total award nominations to count) from his work with the Eagles, Steely Dan, Fleetwood Mac, Sting, John Fogerty, Van Morrison, Toto, Queen, Faith Hill, Lenny Kravitz, Natalie Cole, the Doobie Brothers, Aerosmith, Phil Collins, Aretha Franklin, Barbra Streisand, and many, many others, Elliot has long been recognized for his pristine mixes.

Do you have a philosophy about mixing?

I've always believed that if someone has recorded all this information, then they want it to be heard. Back when I started, if you were able to hear every single instrument in a mix, that was considered a huge achievement. Granted, there wasn't as much information when I started as there is now. I've come across sessions that are a hundred and some odd tracks wide, so it's not as easy for everything to be heard in a track like that.

I have to admit that the way some people record things today is a bit peculiar. All of a sudden you'll be dealing with seven or eight different

mics on the same instrument. It's mind boggling that you have to listen to every single channel to decide which one you want to use. If you pick the wrong ones, they come back at you and say, "Oh, we had a different combination," or "It doesn't sound quite right to us," but they don't tell you what they did in the first place! Granted, mixes can be a little more difficult to deal with because of those issues today, but I still take the same approach of having everything heard with every mix.

If you have a hundred tracks, will you try to have them all heard, or do you go in and do some subtractive mixing?

Well, it depends if that's necessary. I don't usually get those kind of calls where they say, "Here's a hundred tracks. Delete what you want." Usually I'll get between 24 and 48 tracks, and hardly am I ever given the liberty to take some of them out. I just assume that whatever an artist and producer sends me is kind of written in stone. They've recorded it, and unless they tell me otherwise, I usually don't do subtractive mixing.

How often do you work at home?

That's where I mix almost all of the time these days because people don't want to pay to mix in a commercial studio.

You're not mixing in the box at home, are you?

No. I've got a Yamaha DM2000 digital console and a bunch of outboard gear as well.

How long does it take you to do a mix on average?

Depending on how complicated it is, it usually takes anywhere from three hours to a full day.

Three hours is really fast!

Well, a lot of time you just get a vibe and a feel for something, and it comes together, and then you look at it and say, “How much am I actually going to improve this mix?” If it feels great and sounds great, then I’m a little reluctant to beat it into the ground.

For me it’s still about a vibe, and if I can get things to sound good and have a vibe, that’s all I really care about. I still put [legendary engineer] Al Schmitt on a pedestal. He can do three songs in a day, and they’ll be perfect and amazing sounding and have the right vibe, so it’s not like it can’t be done. Some people say that you can’t get a mix in a short time, and that’s just not true; Al’s my proof.

Where do you usually start your mix from?

Out of force of habit, if there’s a rhythm section I’ll usually start with the drums and then move to the bass and just work it up. Once the rhythm section is set, I’ll move on to everything else and end with vocals.

How much EQ do you use?

I can’t say that there are any rules for that. I can’t say that I’ve ever mixed anything that Al has recorded, but if I did I probably wouldn’t

have to use any on it. With some of the stuff done by some of the younger kids, I get it and go, "What were they listening to when they recorded this?" In those cases I'll have to use drastic amounts where I'll be double compressing and double-EQing—all kinds of stuff in order to get something to sound good. I never used to have to do that. Obviously those mixes are the ones that take a day or more.

Do you have any go-to plug-ins that you use?

The only plug-ins I use are the UAD plug-ins. If I need to fix something, then I'll use some of the iZotope plug-ins, but I generally don't use too many. To this day I've never utilized a reverb inside the box. I just don't like the way they sound, so I'm still going to external devices for reverb. My favorite is the [Lexicon] PCM 96. I'm still using a lot of the reverbs built into the desk. I do use the EQs in Nuendo extensively when I need it. I love their EQ.

When you're setting up a mix, do you always have a certain set of effects, like a couple of reverbs and delays, ready to use, or do you patch it as you go?

Usually I don't start out with any reverbs. I'm not one for processing. I'd like to believe that music can survive without reverbs and without delays and without effects.

Obviously when it's called for I'll use it, but the stuff I do is pretty dry. The '70s were a pretty dry time, and then the '80s effects became overused. There was just tons of reverb on everything.

Most of your Steely Dan stuff you did is pretty dry, isn't it?

It’s pretty much dry. If we used anything, we usually used plates.

In the days when I was working at A&R [the now-defunct New York City studio owned by producer Phil Ramone], we had no remotes on any of our plates there. Phil wanted to make changing them difficult because he tuned them himself and he really didn’t want anybody to screw with them. There would be at least four plates in every room. Some of them might be a little shorter than another, but generally they were in the 2- to 2 1/2-second area. There was always an analog tape pre-delay, usually at 15 ips, going into the plates. The plates were tuned so brilliantly that it didn’t become a noticeable effect. It was just a part of the instrument or part of the music. You could actually have a fair amount on an instrument and you just wouldn’t notice it.

How much compression do you use?

Very, very little if I can, but it depends on the project. Growing up at A&R, there were only four pieces of outboard gear in every room: a pair of Fairchild 670 compressors and 2 Pultecs [EQs], and that was it. I used a Fairchild for the bass and the kick drum. There was no EQ on the consoles because we used broadcast consoles, so you were very selective about what you put that outboard gear on. It was usually something where you had trouble getting a sound. It was more about moving the mic or the musician or having them play differently than EQing or compressing.

Most of the compression I did then was hand compression, where you rode a fader and you learned how to ride it smoothly so you didn’t screw up. I never got into the habit of relying on compressors for anything. I probably do a little more now than I did in the past. I use a stereo compressor across the buss usually when I’m mixing, but it’s not for compression as much as it is for the sound of the compressor.

What do you use?

I use a Neve 33609. There's very little of it going on; the needle barely moves, but I like what it does to the sound. Sometimes I'll put a GML EQ across it as well just to add a little air on the very top.

What do you use for monitors and how loud do you listen?

I'm still using NS10s for stereo and five of them for surround. Generally when I'm mixing, I monitor at about 40dB.

Wow, that's quiet.

Yeah, I just find that I can definitely hear the balances better at a lower level so I can hear if I've got a little too much or not enough of something. That comes from wanting to hear every instrument. If I can hear it when it's soft, then it's probably going to be there at any level. I'd like to believe that I can still hear a little bit now because I've monitored so softly for so long.

Do you go up on bigger speakers to get your low end?

No, I might go up to 80 or 85dB when I'm putting the kick and the bass in just to hear where they're sitting comparatively, but once I establish that, I'll go right down to 40 or so.

Do you have any listening tricks, such as going outside the control room or anything like that?

I used to take a mix home to listen on my speakers there, but then I found out that I was getting better results by listening in whatever car I was driving. A lot of guys used to base their mixes on what it sounded like in a car because a lot of the times the car was the lowest common denominator.

Usually just two or three. Pretty much vocal up and vocal down unless there's something so weird that I'm worried about it; then I might do an alternate version with something lowered or out. I've got to say that with most engineers, your initial instinct is usually correct.

Do you draw your automation in on the workstation or do you only use fader automation?

I use the console automation as my primary automation, but occasionally I get more tracks than I have outputs on the DAW so I'll combine tracks inside the box and use the automation in Nuendo to do whatever needs to be done.

Do you have any special tricks that you use when you're mixing?

I've never relied on any trick stuff. I wish I could say that I've been using something all these years that's some kind of secret, but I don't do anything that's of a secret nature. I'm more than willing to tell anyone who asks me what I did because I've never had any tricks. I've always felt that the real trick is in how you hear it.

Let's talk about surround, since you've done so many surround mixes. Has your approach changed over the years?

In some regards, yes, but as far as the overall approach, no. I still do a very aggressive mix. I like as much information coming out of the surrounds as I do the front, so I'm still as aggressive as I was when I first started, maybe even more so. The only thing that's really changed for me is how I use the center speaker. I try to use it a little more than I have in the past. I've found that when listening in a car, the center speaker is a little more important than it is in the home. If I put a vocal in there, it's going to be at least as loud as the phantom center, maybe a little more.

Do you only put the lead vocal in the center?

No, I put the snare, bass, kick drum, and sometimes an instrument that's a mono track, like a tambourine or bells or something that I can just put in one speaker.

Do you still keep the mix fairly dry?

Yeah, pretty much.

Do you use the LFE much?

I run the speakers full range so I don't have to put all that much in the LFE. I always worry about bass management in people's homes, since every receiver does it differently and there's no way to predict what will happen, so I just ignore it and put a minimum of stuff into the LFE.

What's your approach to doing a live concert as compared to a studio record in surround?

I'm probably the only guy who does it, but I don't believe in setting an arena perspective in surround. I don't like the idea of having just some ambience and room mics in the back. I still want to hear the music coming from everywhere. I still try to be as aggressive with live stuff as with a studio record, so I take the creative license to move to the stage and put instruments in the rear.

I think if the rear speakers aren't lit up, the listener feels cheated.

Yeah, I agree with you. What really gets me off is hearing information coming from the rears because that's what's unusual about surround. We've been listening to the front side for as long as we've had recordings. Now that we have other speakers, that's where a good deal of the information should be coming from.

Would you have any advice for someone who's just starting to mix?

I would say that you have to believe in yourself. You can't second-guess what you're doing. I've always been of the mind that if I can make myself happy listening to a mix, then hopefully the people who are employing me will be just as happy.

I don't try to guess what someone might want. If there's someone there in the room with me when I start a mix, I know that sooner or later I'm going to hear whether they hate it or they love it, but generally I try to mix for myself. At this point in my career I know that if people are calling me, then they must like what I do. Just remember that what we do is to convey the artist's feelings and make it as musical as possible without harming it.

Andrew Scheps

Andrew Scheps has worked on mega-hit albums for a who’s-who of superstar artists such as Red Hot Chili Peppers, Metallica, U2, Justin Timberlake, Jay-Z, the Rolling Stones, Linkin Park, Jewel, Neil Diamond, and Adele.

Even though he’s working out of his pretty outstanding home studio built around dual Neve 8068s, a massive wall of outboard gear, and dual Studer A800 24-track tape machines, amazingly Andrew is not one still living in the analog past, as the DAW is an integral part of his workflow. To find out more about Andrew and to see pictures of his great studio, go to punkerpad.com.

Can you hear the final product in your head before you mix?

If I know the song, then I already have a pretty clear picture of what I’d like it to be. If not, I’ll usually get that the first time I listen through a track. It’s not so much for the sonics, but more in terms of size, like figuring out how big the chorus will be. Sometimes I’ll get really specific ideas about effects that I’ll try as well.

In terms of starting a mix, I think the main thing, especially if it’s a song I haven’t recorded, is that I go through instrument by instrument to see how it sounds, but what I’m really doing is learning every single part so that when I come to build my balance, I know where everything is going to be.

Do you have a template for your effects before you start to mix?

Kind of, although I don't use a lot of effects. I use a lot of parallel compression so that's more of what I have set up. In terms of what gets sent to those compressors, some of it is consistent and some of it changes with every mix, but they're ready for me at the push of a button, which on an analog console is great because I just leave that part of the patchbay alone.

In terms of effects, sometimes I'll have one kind of chorus-spreader kind of thing and one reverb and that's it. I don't tend to use many effects because a lot of the stuff I mix is straight-up guitar rock, and it's more about the balance and making things explode.

Do have an approach to doing that?

You're never really as aware of your own process as you think you are. I'll think that I really didn't do much of anything, and then I'll look at a mix and find that I'm using 50 things on it.

Also, because I mix on a console there's the whole process of laying out the outputs of Pro Tools to see where everything is going to come up on the console. There are things that always live in the same place, like channel 24 is always the vocal, so I'm usually figuring out how to lay out everything between the drums and the vocal. I do that while I'm finding out what everything is doing, so there's a long discovery process where it doesn't seem like I'm getting much done, but then everything happens really quickly after that.

Where do you build your mix from?

It depends. I'd love to say that I always build it from the vocal, but usually what I'll do is deal with the drums to get them to act like one fader's worth of stuff instead of 20 or whatever it is. Once I've gone through that process that I just described, everything seems to come up at once. I'll have listened to vocal and the background vocals and know exactly where they are, but I'll get the band to work without the vocals first, which I know a lot of people don't think is a good idea.

I think it's the same thing when you're working on a particular instrument in solo. After 20 years, my brain sometimes unconsciously knows what an instrument will sound like soloed, so I'll tend to get the tone on things separately, and then it's all about the balance. I almost never have to go back and change things once I get the vocals in. My brain seems to know what that balance is going be when the vocals are inserted.

How much do you do in the box?

I always think that I do nothing in the box, but I really do a lot of the technical things. The EQs on the Neve are very broad and very musical; they're not good for anything surgical. If there's a nasty frequency in the overheads or the snare is ringing too much, I take care of all of that in Pro Tools. Usually I'll have the background vocals coming out of one stereo output pair, so I'll deal with them in the box. Sometimes I might split a couple of them out, but I don't want 20 tracks of background vocals on the console; it's just a waste. A lot of the crazier effects can come from plug- ins there as well.

There's quite a bit that goes on in Pro Tools, but it's more about shaping things before they get out into the console. The console is much more of an organic balance thing, while Pro Tools is more for making

things sound the way I want them to sound. The console is more about putting it all back together and mixing it.

I actually mixed in the box for years in this same room. I had a [Digidesign] ProControl in here, and that was great. In fact, there are some things that I mixed in the box that I listen to now and go, "Wow, that sounds really good."

I don't have any philosophical differences with mixing one way or the other way. It's more of once you have the console, as much of a drag as it is to document everything, it's such a joy to mix on it. When I'm mixing, it doesn't matter whether it's coming off tape or Pro Tools; it's just faders and speakers, and that's it. I love that because sometimes mixing in the box makes you so precise that you then fix things that don't really need fixing. I like the sloppiness of doing it on the console.

Do you find that you're using your outboard gear less?

No, not at all. When I document every mix, I wish that was the case because it's a lot more to write down, but because a lot of it is parallel processing and stays patched in, it's so much faster for me to hit a button on the console than it is for me to set the same thing up in Pro Tools. I may send the bass, the guitars, and the background vocals to a stereo compressor, and in doing that in the box, it could change the balance on the board, so that doesn't really work for me at all. It's less of a sonic thing than a convenience thing.

What do you use the parallel processing on?

On this mix right now there's a parallel compressor on the kick and snare, then there's another just on the snare. There's a stereo one on the toms and overheads, a mono one on just the dirty bass (this song has three basses), a stereo one on the guitar and vocals, and then a couple of different ones just for the lead vocal—one that's sort of spitty and grainy and one that's sort of fat. That changes from mix to mix. In fact, it changes a lot.

Are you tucking the parallel processed channel just underneath the unaffected one?

Yeah, although sometimes the parallel one ends up being pretty loud, in which case it's almost like using an insert compressor, but it's across a few things. Sometimes it's just tucking it in to add power or weight.

Do you EQ the parallel processing as well?

Not much, because everything is post-fader, so it's only the EQ'd stuff that gets in there. Sometimes on the drums that will start to really bring out a badly ringing cymbal, so I'll go in and do something surgical to fix the problem, but most of the time it's the same tone as the uncompressed.

Are you using buss compression as well?

Yeah. I used to never compress the mix because I could never find one that I liked that didn't take away from everything else that was going on. Then a couple of years ago I started using the 2264s [the onboard Neve compressor modules] that are in the console. I also learned the lesson that if you're going to compress the mix, you begin your mix with the compressor already on. You don't get your mix and then put your compressor on, because that doesn't work. You have to mix to it.

I don’t use heavy compression, though. I don’t think I ever add more than 3 or 4dB, so I’m not really smashing it. It is a pretty aggressive setting, though, like a super-fast release. I’ve tried printing uncompressed mixes and bringing those to mastering, but you can never re-create the sound, so I always mix to it now.

How many alternate versions of mixes do you do?

If there’s nothing specific that comes up during the mix, I’ll do a full mix, an instrumental, sometimes a TV mix just because people are used to getting it, a vocal up and a vocal down, and an a cappella. If there’s been a lot of talk about the balance of the backgrounds, I’ll also give them an a cappella lead and an a cappella background. If there’s been talk about any particular instrument that the band doesn’t agree about, then I’ll print them an alternate of the other variation they’re considering and then strongly label the one that we think is the right one.

How long does it take you to do a mix?

It really depends upon the material. If it’s well recorded and I’ve already done a song for the album, then it can come together in as little as three or four hours. The first one on an album usually takes longer because there’s a lot to sort out, so it will usually take a full day. I’m usually ready to play something for the band by late afternoon at the latest.

What are you using for monitors?

I love the old Tannoy SRM-10Bs. One of the first people I worked extensively with had these speakers; then I borrowed a pair for a gig and just fell in love with them. Since then I've tried seven or eight times to find something else to use beside them, but their midrange and top end are different from most other speakers, which makes it really hard to go back and forth. Now that I work mostly in my own room, I've gotten to the point where I don't have to listen on anything else. I never switch speakers and I never listen in the car when I'm mixing, yet almost never does anyone say, "Ah, there's a problem with the low end." At this point I own three pairs here and another pair in Europe, and that's all I'll ever use because I know that I can walk into any room and be safe with what I'm hearing. I power them with an old Crown DC-300A, and they just match well. Whenever I take the speakers somewhere else, though, I just use the amp they've got, and it's usually fine.

How loud do you monitor?

When I'm getting the mix together, I monitor pretty loud and for longer than I probably should. Once I get the balance, then I mix really quietly, but still occasionally check things loud every once in a while. Mixing is such an emotional thing. You're trying to get it to seem exciting, especially on the rock stuff, so you have to hear it loud to know that the kick and the snare and vocal are hitting you in the chest.

There are things that you can't judge when it's loud, though. You can't judge the vocal level properly because the vocal will sink into the mix more when it's loud, but in terms of impact and emotion, you've got to crank it.

You said that you mixed in the box for a long time. How did you get back into using a console?

For Stadium Arcadium [by Red Hot Chili Peppers], the band wanted the record mixed on a Neve. It was tracked on a Neve, and they wanted it mixed on one, too. I tried all of the available Neve rooms in town, and it just wasn't working for whatever reason, so I ended up renting a console and realized that this room could easily accommodate it. The other thing is that I just loved having a console, and as an investment, it's not going down in value—unlike an SSL, whose values are still plummeting.

The first desk had 32 inputs, but when I knew that I was going to be mixing Metallica, I knew that I needed more. Thirty-two inputs plus the other 10 on the BCM-10 [the small Neve broadcast desk that Andrew uses as a sidecar console] wasn't going to cut it, since their drums alone had around 30 channels. They weren't all going to be coming up on the board, but 32 just wasn't enough inputs. I was going to rent the console that I used for Stadium Arcadium again, but in the process convinced them to sell it to me. As far as the outboard gear, studios keep closing, and that's where most of that comes from, since it tends to come in batches.

Where do you do your automation—in the box or on the desk?

Most of the automation is done on the desk. The only thing done in the box is for extreme fixes. Once the mix is pretty much done and we're adjusting something like background levels with the band, it's easier to do that in the box because sometimes it's just a certain word that they want louder, and then you don't want to be sloppy with a fader ride on the console. It's more precise in the box.

Do you ride the rhythm section for fills or is your mix fairly static?

I still do ride things because the compressors are sucking out some of the natural dynamics of how the instruments were played, especially on some of the louder rock stuff. I'm only adding it back in, rather than creating something out of nothing. I don't turn the whole mix up at every chorus, for instance. Some people can do that with success, but I always hear it when I do it. I do definitely push the drums for the downbeat of the chorus and really try to accentuate anything that might be cool in a guitar performance, as well as some of the idiosyncrasies. The rides aren't drastic, but most things are moving.

Ken Scott

Legendary producer Ken Scott began his career at the Abbey Road Studios working with the Beatles on The White Album and Magical Mystery Tour; on six David Bowie albums, including the seminal Ziggy Stardust album; and with Pink Floyd, Elton John, Duran Duran, Jeff Beck, Supertramp, Procol Harum, Devo, Kansas, Mahavishnu Orchestra, and many more. To put it mildly, he's an absolute icon in the recording industry, having been a part of records that have conservatively sold more than 200 million units.

I got to know Ken while co-writing his memoir, Abbey Road to Ziggy Stardust, which is filled with stories about him working with some of the greatest artists in the world, as well as a lot of technical bits that engineers love. During the writing of the book, I was scheduled to produce the third album for the band SNEW, and I sheepishly asked Ken if he would track the basics for me. Much to my and the band's delight he agreed, which was the beginning of a great experience watching a man of such legendary status work.

Ken was a study in quality via fundamentals and minimalism. While many engineers wheel in a number of tall racks filled with pricey and exotic gear, Ken used nothing more than what the studio had. During tracking he occasionally used a compressor on the bass, guitar, or vocal, but that was all.

The sounds he got were almost instantly wonderful. A little EQ here and there (and I mean a little), and we were all blown away. It reminded me

of how important fundamentals really are and how easy it is to get away from them.

When it came time to mix, I had some pretty good roughs that everyone was happy with, but we all wanted to work with Ken again. Once again, much to our surprise and pleasure, he agreed to do it, and once again we got to see and hear just how effective great technique and ears can be. With the help of only a single reverb, a couple of delays, and sometimes a chorus/stereo spreader, Ken did his thing. Occasionally he used a compressor on one of the instruments and just a dB or so of an SSL compressor across the mix buss, and every now and again a plugin in the DAW. I came away thinking, "So this is how he got the sound on those Bowie records." Big ears and experience trumps tons of gear any day, at least for some kinds of music.

I've learned so much from all the mixers featured in this book, but Ken's lessons are the most profound. A mix can sound great without a huge amount of tinkering if your tracks are great to begin with (which they were because he tracked them). Of course, the ears and experience of working on albums that we all hold as legendary helps as well.

While a lot about how Ken works can be found in Abbey Road to Ziggy Stardust, I asked him some mixing questions that we deemed too technical for that book but perfect for this one. To read Ken's vast discography, visit ken-scott.com. To discover more about the Abbey Road To Ziggy Stardust book, visit abbeytoziggy.com.

What's your philosophy about mixing?

It all starts from recording. That's why I don't like mixing other people's stuff—because I find that I can't do as much with it. If I recorded it, there's an inherent thing of everything finding its own place.

I like everything to be heard—not necessarily the first time you hear it, though. If you listen to something a second or third time, you might go, "Wow, I never heard that guitar part back there," but everything has to have its own place. That's probably a question of the EQ and general sounds I use, but I also like a mix to have depth, too, which I do by using distant mics when I'm recording and/or reverb.

The other thing is to make decisions as much as you can along the way. Everyone is too willing to let any decision wait until the last minute instead of making it on the spot. When I started, we had to make the decision on the balance of the instruments or how much reverb we added because we only had a limited number of tracks, but that was good training for later even when we had a lot more tracks to play with. Today you see people piling on track after track and waiting until the mix to sort it out. That makes everything take a lot longer and makes it harder to mix because you're not sure how everything is going to fit together.

I've noticed that there are certain frequencies that you always use. How did that come about?

Because of what I was trained on. The EQ on the EMI desks [back at EMI Studios, where Ken started] was so limited that you tended to get used to certain frequencies. There was 100Hz and 5k on the desk, and then we had an outboard EQ that could give us either 2.7k, 3.5k, or 10k. I usually now EQ somewhere between 4 and 5k, 10k, and more like 60 or 80 these days. Times have changed in that we go for more low end now than we did back then.

The one thing that did change for me over time was my not liking 200Hz. That frequency couldn't be touched in the early days because we didn't have an EQ that was centered there, and it wasn't until later on that I decided that I didn't like it and began to pull it out. I don't know exactly when that happened though, but I know that I was doing it by the time I mixed [Supertramp's] Crime of the Century.

Do you pull 200Hz out on everything?

No, bass drum definitely and sometimes the bass.

I know that records from the '60s and '70s always seemed to be centered around the bass instead of the drums. When did the low end on records begin to change?

Probably with the advent of hi-fi systems at home, and when we finally got the option to EQ at a lower frequency. Once I got to Trident, I started to EQ the lower frequencies because I could, and with the Lockwoods [speaker cabinets with Tannoy speakers in them], we could then both hear it and feel it.

Can you hear the final mix in your head before you start?

No, not the final mix. The general mix, yeah, but quite often I try to get it as close to what I think the final mix is going to be during recording. Stanley Clarke even mentioned about how I had what sounded like the final mix going the entire time we were tracking.

It all comes from the idea that if you don't have the mix going as soon as possible, how do you know what sounds work? You don't know if a

frequency on a keyboard or guitar will work together or if something will mask it. You can never gauge it. That's why people take so long in mixing these days, because they don't make the decision up front what it's going to be like, so they never have the correct mix.

You're pretty minimal with effects.

Once again, that comes from the old days, because I didn't have much to use. What I've found is that the more reverbs you use, the more it tends to pull your attention away from the important things in the song because it's not too natural. If there are many instruments playing in a room, everything naturally has the same ambient sound. If you put a different reverb on each instrument, it starts to sound totally unnatural. If you think about all of the Bowie and Supertramp stuff that I did, I used a maximum of two reverbs, and most of it was only one.

When I used two, it would've been an EMT plate and a Cooper Time Cube (see Figure 30.1), so it was one long and the other short.

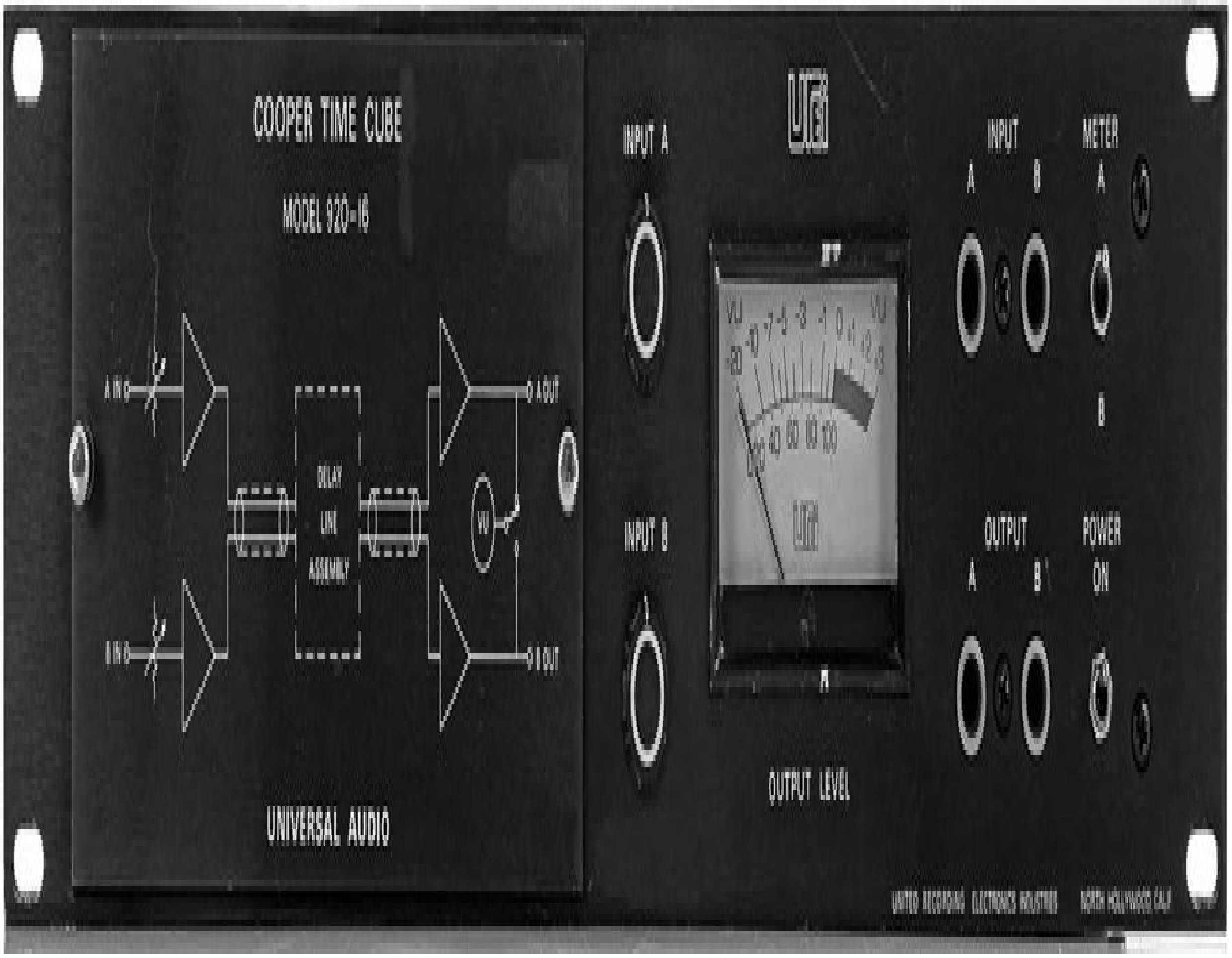

Figure 30.1: A long out-of-production Cooper Time Cube.

How has your workflow changed?

Number one is that I had to change to mixing a song from beginning to end instead of mixing it in sections like I used to. I'd love to go back to doing it one bit at a time. When I'm doing a short section, I can just listen through to all the instruments one at a time and focus on them better. When I'm doing the complete song I might start listening to the bass; then before you know it, I'll be listening to a guitar instead. Back in the early days before automation, it was like an orchestra mixing

because everyone in the room had things to do, and we all had to do them in sync to be able to get the mix happening. Even though I was doing it in sections, within those sections there were things changing, so we needed at least five sets of hands on the board to get it done.

Were the mixes better because of the "all hands on deck" feel?

Yes, they were more organic. Today, the fact that we can get everything as precise as possible leads us to think about the mix too much. Perfection comes from the soul and not from the brain. Because it's all in the computer you think about it more and don't feel it as much. When you get all of those people working together it feels better, although you can't do the same thing on a computer. The same thing with using a lot of reverbs; it becomes less organic.

One of the things I do like to do is get the mix as close as I can with the board automation, then just go in and adjust it a bit in the box. That's very useful.

You're pretty mild when it comes to using compression, even on individual tracks.

Sometimes I do it more than others. It depends upon the desired effect. I tend to limit heavier on acoustic guitars and sometimes the piano, but it depends upon the part and what is required.

Did someone teach you how to mix?

To a certain degree, yes. That consisted of just sitting there and listening to what Norman Smith, Malcolm Addey, and Peter Bown [the EMI staff engineers when Ken started there] did, and then attempting to follow in their footsteps. It's also getting that confidence that what you're hearing in the control room is correct for you. It's all about the confidence in the monitors for me. If I'm doing a lot of work in a new place, I'll take the first part of the first session and just listen to stuff to get an idea of what it's like.

I noticed that you don't listen on the small speakers much.

I like to work mostly on the big speakers, but I always check things small. I used to check on Auratones, but these days it will be NS10s or something like that. When I work with people who bring their own set of speakers, it just gets more and more confusing.

Every speaker in every room is going to sound different. I can use almost any studio as long as the monitors are good, because then at least you have the confidence in what you're hearing. It doesn't matter whether you're using crappy mics; if you're getting a sound that you like coming off the monitors, then you know it works. If the monitors are off, you have no idea about anything. You can use the most expensive gear in the world, and it can still sound bad.

You listen pretty loudly, don't you?

Do I? [Laughs] When you're working every day, you tend to start to turn it up louder and louder. I know that at the moment I don't monitor as loud as I used to, but it's because I've spent more time working on my book than being in the studio lately, so my hearing's become more

“normal.” If you’re in there day after day, project after project, you tend to want it louder and louder.

Did you ever bring gear with you to the studio?

No, the only things that I ever brought were my guitars, but never anything else.

What would you suggest to become a better mixer?

First of all, always believe that you can get better. The hardest time that I ever had was when I thought I’d achieved perfection with Crime of the Century. It wasn’t until I started to find fault with it that I could move on. You should always be learning from everything that you do.

These days I can’t necessarily say that it’s me getting better, but I’m always striving for what I’m working on to be better than I’ve done before. Plus, you have to be accepting of the situation under which you’re working. If you don’t always have a guitar virtuoso to work with, don’t expect to get that kind of sound or performance out of the guitarist you’re working with. You can expect to get the best out of the player and have it add to the overall sound, though.

Is there one instrument that you take more care with than anything else?

Whatever I determine to be the most important thing of the recording, that’s what I will spend the most time on or concentrate on. There are some things that I can’t stand the mix of today, but it was exactly what we wanted to do at the time and was agreed to by all concerned. It’s purely dependent upon the finished product every time.

Ed Seay

Getting his starting in Atlanta in the 70s engineering and producing hits for Paul Davis, Peabo Bryson, and Melissa Manchester, Ed Seay has become one of the most respected engineers in Nashville since moving there in 1984. With hit-making clients such as Blake Shelton, Lee Brice, Martina McBride, Ricky Skaggs, Dolly Parton, Pam Tillis, Highway 101, Collin Raye, and a host of others, Ed has led the charge in changing the recording approach in Nashville. In this updated interview, Ed also describes how both he and Nashville have embraced mixing "in the box."

Do you hear the final product in your head before you begin to mix?

To some extent I can. Rather than just randomly pushing up faders and saying, "Well, a little of this EQ or effect might be nice," I like to have a vision as far as where we're going and what's the perspective. Definitely, I try to grasp that early on.

Is there a difference between mixing country music and other genres?

Country music is definitely lyric driven. One of the mistakes that some people make when they try to work on the stuff is they tend to downplay the lyric or downplay the lead vocal. In pop and in rock, sometimes you don't always hear every word, and it's kind of okay if it's buried just a little bit, but country is usually not that way. People definitely sing along with country songs, so that's the biggest thing. The vocal rules, but at the same time, it's pretty boring if it's all vocals and it sounds like a country record from the '60s, where you don't have any

power in there. There's an art to keeping the vocal on top without making it dominate.

How much do you mix in the box?

I mix in the box about 95 percent of the time. If someone says, "I wanna mix on a big console," I say, "Great. Let's go," but every time I do that I look back and think, "That sure is easier to do in the box." Actually I started mixing in the box back in 1999, so I was one of the pioneers, at least in Nashville. I saw what was coming and embraced it not only as the future, but also as the present. In fact, I mixed the first in-the-box country record to go number one: "Austin" by Blake Shelton [in 2001].

Part of what's really changed in the business recently is that we live in a recall world now. It's so important to have the ability to recall something 10 times to turn something up or down, and that gets cost-prohibitive on a console since your paying for the room and the assistant and everything that comes with a studio. Mixing in the box is really the way records are made these days, and so many of the big records are done that way. I'm glad that I embraced it early.

When you start to mix, how do you build it?

Well, I'll usually go through and push up instruments to see if there are any trouble spots. All this is dependent upon whether it's something that I've recorded or if I'm hearing it fresh and have no idea what it is. If that's the case, then what I'll do is kind of rough mix it out real quick. I'll push it up and see where it's going before I start diving in.

If it's something that I know, then I'll go through and mold the sounds in a minor way to fit the modern profile that it needs to be in. In other words, if it has a real flabby, dull kick drum, it doesn't matter what the vision is; this kick drum's never going to get there, and I'll do whatever I have to do to make it so. I'll work through my mix like that and try to get everything up into the acceptable range, or the exceptional range, or at least somewhere that can be worked with. It takes a couple of hours to get good sounds on everything, and then another couple of hours to get real good balances. After that I'll do some frequency juggling so that everybody is out of everybody else's way.

The last stage of the mix and the toughest part is the several hours it takes me to make it sound emotional and urgent and exciting so that it's just not a song, it's a record. It's taking it beyond sounding just good and making it sound like an event.

How do you go about doing that?

I try to find what's important in the mix. I try to find out if the lead vocal is incredibly passionate and then make sure that the spotlight shines on it. Or if the acoustics are sitting there but not really driving the song like they need to, sometimes playing with compression on them can make it sound like, "Boy, this guy was into it." Maybe it's pushing and pulling different instruments. Somebody's got to be back, and sometimes it's better when things are back and other things are farther up front. Sometimes it means making sure your cymbals or your room mics are where you can actually feel the guy, or sometimes adding compression can be the answer to making the thing come alive. Sometimes hearing the singer breathe like on the old Steve Miller records. With a little of that, you might say, "Man, he's working. I believe it." It's a little subconscious thing, but sometimes that can help. It's just basically playing with it and trying to put into it that indefinable thing that makes it exciting.

When you're building your mix, are you starting with bass first or starting with the kick drum?

I start with the kick drum sound, but then I put up the drum kit and put the bass in. Then I'll push up all the static channels that aren't going to have giant moves, like the acoustic stuff, keyboard pads, maybe a synth or Rhodes or piano that doesn't have a whole bunch of stepping-out licks.

Early on, I'll try to make sure that there's room for the lead vocal. I think one of the big mistakes is to work on your track for eight hours and get it blistering hot and barking, but then have no way for this vocal to cut through. You're then faced with the choice of turning this baritone vocal into steel wool with ridiculous EQ, or just turning him up so loud that he sounds inappropriate. It's cool to have a bright record as long as everything kind of comes up together, but if you've got an incredibly bright snare drum and the vocal's not so bright, then it makes the vocal sound even duller. If you're thinking all the way to the end when you master the record and add EQ, it'll brighten the vocal, but it's also going to bring up the snare even more, so you have to have everything in perspective.

Eventually I get the vocals in and get the backgrounds around them; then I put up the solos and the signature stuff. At that point I get an overall rough balance of everything that sits there pretty well and then juggle the pieces.

What's your approach to EQ?

I just try to get stuff to sound natural, but at the same time be very vivid. I break it down into roughly three areas: mids, the top, and the bottom; then there's low mids and high mids. Generally, except for a very few instruments or a few microphones, cutting flat doesn't sound good to most people's ears, so I'll say, "Well, if this is a state-of-the-art preamp and a great mic and it doesn't sound that great to me, why?" Well, the midrange is not quite vivid enough. Okay, we'll look at the 3k, 4k range, maybe 2500. Why don't we make it kind of come to life like a shot of cappuccino and open it up a little bit?

Then maybe I'm not hearing the air around things, so let's go up to 10k or 15k and just bump it up a little bit and see if we can kind of perk it up. Now all that sounds good, but our bottom is kind of undefined. We don't have any meat down there. Well, let's sweep through and see what helps the low end. Sometimes, depending on different instruments, a hundred cycles can do wonders for some instruments. Sometimes you need to dip out at 400 cycles, because that's the area that sometimes just clouds up and takes the clarity away, but a lot of times adding a little 400 can fatten things up.

On a vocal sometimes I think, "Does this vocal need a diet plan? Does he need to lose some flab down there?" Sometimes we need some weight on this guy, so let's add some 300 cycles and make him sound a little more important. It's kind of contouring.

Also, frequency juggling is important. One of the biggest compliments people give me is that they say, "You know, Ed, on your mixes, I can hear everything." There are two reasons for that. One is I've pushed things up at the right time when they want to hear it, but the other thing is I don't EQ everything in the same place. You don't EQ 3k on the vocal and the guitar and the bass and the synth and the piano, because then you have such a buildup there that you have a frequency

war going on. Sometimes you can say, “Well, the piano doesn’t need 3k, so let’s go lower or let’s go higher.” Or, “This vocal will pop through if we shine the light not in his nose, but maybe toward his forehead.” In so doing, you can make things audible, and everybody can get some camera time.

Do you have a specific approach to panning?

Yeah, I do. The most significant approach is I pan as if I were sitting in the audience, especially with the drums. The reason is, I don’t play the drums; therefore, I sit in the audience and listen, and that means with most drummers their hi-hat is to the right (unless they’re left-handed). To me, I can get away with anything except the drums being backwards, because it just strikes me funny. However, I thrash a bit at piano, so I always put the low end on the left-hand side and the high end on the right-hand side.

Hard left and hard right?

Usually, but not always. With a piano, it depends on how phase coherent it was recorded. If it’s not dramatic stereo, I’ll try to make it more dramatic. Also, if whoever recorded it didn’t pay really good attention to the phasing on the mics and the thing is way wide and it falls apart in mono, I’ll be panning it in so that in mono it doesn’t go away. Sometimes flipping the phase on one side can fix that because a lot of people don’t check. Of course, stereo is more important now than ever before, but on a lot of the video channels, you still might be listening in mono, so I check for that.

I try to make stereo records, and I’m not afraid to pan something extremely wide. I like my mixes to have a few things that stick out and

get some attention and not just blend in with the crowd, so I always put the electric guitar on the left and steel on the right. That way, there can be all kinds of contrast—not only volume dynamics, but panning dynamics as well.

One of the things I don't like is what I call "big mono," where there's no difference in the left and the right other than a little warble. If you pan that wide left and right, and then here comes another keyboard and you pan that left and right wide, and then there's the two guitars and you pan them left and right wide, by the time you get all this stuff left and right wide, there's really no stereo in the sound. It's like having a big mono record, and it's just not really aurally gratifying. To me, it's better to have some segregation, and that's one of the ways I try to make everything heard in the mixes. Give everybody a place on the stage.

Do you use compression as an effect, or just to even things out, or both?

Both. To me, the key to compression is that it makes the instrument sound like it's being turned up, not being turned down. If you choose the wrong compressor or you use it the wrong way, then your stuff can sound like it's always going away from you. If you use the correct compressor for the job, you can make it sound like, "Man, these guys are coming at you." It's very active and aggressive. Quite often, I'll use it on the stereo buss, but I try not to be too crazy with it.

If you remove all dynamics or if you really lean on it in an improper way during mixing, when it goes to mastering there's not much for the guy to do there. If he does, it'll only compound the problem; then by the time it gets on the radio there's nothing left that'll pump the radio

compressors, so then it just kind of lies there. It's loud, but nothing ever really jumps out of your mix.

But yeah, lead vocals almost always get compressed. Bass, certainly when I'm tracking it and quite often when I'm mixing it. I time the release to the tempo of the song or to the end of the note release, especially if the guy's using flat wound strings for more of a retro bass that has a lot of attack and less hang time. Sometimes if you use the wrong compressor on a snare drum, all you'll get is the initial hit and then it'll turn it down, but if you use the right kind of compression, slow the attack down, speed up the release, you'll get a different effect where there's more length to the snare sound. It'll come sucking back at you. Compression's important, but it's gotta be the right kind, and I think that's the key.

Are you compressing more because of the current tastes of the business?

Country records are a lot more aggressive sounding these days, and that comes from the compression. I don't know if I'm compressing more, but I'm compressing smarter. Rather than piling it all onto one master compressor and flat-topping it to get the volume there, I'm trying to do smarter compression along the way. It can be just as loud and maybe even more exciting, but it doesn't just sound like the guy in the rocket sled with his face pinned back.

I'm using less compression on individual instruments but more on the vocal. Usually low ratio compression where it packs it up gently and you say, "Man, that guy can really work the mic." I'm also doing more intelligent compression on the mix buss.

Ultimately, when you send something out to a client, if you don't crank the level up to at least a level close to what he's used to hearing, you'll lose the gig. When it comes time to deliver it to mastering, I'll drop the level back and leave the mastering engineer some room to work with while retaining the sound that the client has learned to love.

What are you using on the mix buss?

An [Waves] L2 is good. It's kind of gain with no vibe, while the [Waves] L3 is gain with a vibe. The Slate FX-G is gain with vibe and color, where you can let different things poke through using its controls.

Do you add effects as you go along, or do you get the balance up and then add the effects?

I kind of come and go with this. What I'll do is try to make things sound as good as I can dry. If I hear something that just sounds too iso'ed [isolated] and too unrelated to the mix, then I'll add some effects as I go, but I don't go too crazy with it until I get the whole picture. Once it's all sitting there, you can easily tell if it's just not gluing together. My general setup for a mix is I'll have one send set up for long verb and another set up for a short, kind of a room simulation.

Long being what, 2.5, 3 seconds?

Yeah, 2.5, 2.3. For a ballad, sometimes 2.6. Then I'll usually have a delay send with eighth note or sixteenth note or dotted eighth triplets. Sometimes I'll have a little pitch change going. I may have a gated reverb or something that can kind of pull sounds together. Sometimes an isolated guitar sounds great dry and in your face by itself, but other times it seems like, "Wow, they had an overdub party, and look who

showed up." Sometimes a little of that gated reverb can kind of smear it together and make it sound like he was actually on the floor with them.

Aren't most country mixes still on the dry side?

Things are still dry, but I worked with a famous country artist who said, "I miss reverb." [Laughs] I'm more into contrast now, where the record may be pretty dry, but you get this long United Western chamber from Altiverb (the same one that Sinatra and the Beach Boys used) wash on some things. It's a sound, and it puts you in a place.

Some things aren't meant to be all dry, and some things are. Sometimes you just use enough that when you take it away, you go, "Oh, what just happened?" but you don't really notice it otherwise. It'll come full circle one of these days when it will be cool to be wet again.

Are there any go-to plugins that you always use?

As far as effects, one of my favorites is the Reverb 1 [the native Pro Tools plugin], and another I use a lot is the [Audio Ease] Altiverb. [Trillium Lane Labs] TL Space is another that I like, which is similar to the Altiverb. As far as compression, I like the CB 303 by McDSP, which is brilliant. For EQs I like the Massenburg DesignWorks five-band EQ and the seven-band EQ3 by Avid. The thing I like about that is the meter is very accurate. If you don't see red, it's not distorting, unlike some other plugins. It's so good that I'll sometimes put it first in line on my master fader just for the metering.

What do you use for monitoring?

I don't change monitors very often, but for my bigs I now have a Carl Tatz Design PhantomFocus System, which is great. What I love is that I can tell what I've got from DC to radar, and the imaging is so accurate.

I love sitting in front of Carl's monitors—or any large monitors, really—but that's not always a real-world situation. At least 60 percent of the time, I'm off on the side listening to a pair of passive Von Schweikert monitors, which are a little bigger than NS10s. They're sitting off to the side because it's so obvious to hear the balances when you're not in the sweet spot sometimes. They're great for hearing balance and tonality more like the average listener is going to hear it. I still have a boom box that I check every now and then.

Of course, it still amazes me that with all these great and expensive tools, I can still throw it in the car, and in like two seconds I immediately know what's wrong. What I like to do is make it sound good on all three unrelated systems; then it's going to relate to the rest of the world.

How loud do you usually listen when you're mixing?

I mix at different levels. I try not to mix too loud because it'll wear you down and fool your perspective. Sometimes it's very valuable to turn things down, but there's an up and down side to it. If you listen too soft, you'll add too much bass. If you listen too loud, you'll turn the lead vocals down too much.

How many versions of a mix do you do?

People live with a song for so long these days that by the time it gets to the end they pretty much know how it should be. As a result, I'll print a mix and maybe a vocal-up version, but I don't do too many versions since I can get one out almost instantly if someone wants it. Occasionally someone will ask for a version with less reverb or something like that, but alternate mixes are almost a non-issue anymore.

Are there any tricks you've developed to give you a console sound?

I've been a fan of the Crane Song Phoenix plugin for a long time, and as of late I like the Slate Audio Virtual Console Collection, which gives you different console sounds. It kind of pulls things together in an analog way. Is it exactly the same as a console? No, but it's not better or worse; it's just different. When it comes right down to it, it's still the driver more than the car in most cases.

Glossary

0 dB Full Scale. Abbreviated FS, it’s the highest level that can be recorded in the digital domain. Recording beyond 0dB FS results in severe distortion.

5.1. A speaker system that uses three speakers across the front and two stereo speakers in the rear, along with a subwoofer.

808. One of the early drum machines made by Roland favored for many year in hip-hop and EDM.

air. Frequencies above 10kHz that are more felt than heard. These frequencies can provide more realism to a sound used in the correct proportion.

airplay. When a song gets played on the radio.

ambience. The background noise of an environment.

arpeggio. The notes of a chord played in quick succession.

arrangement. The way the instruments are combined in a song.

articulations. The way a note or phrase is played or sung in terms of attack, release and duration.

Atmos (see Dolby Atmos)

attack. The first part of a sound. On a compressor/limiter, a control that affects how that device will respond to the attack of a sound.

attenuation. A decrease in level.

attenuation pad (sometimes just called a pad). A circuit that decreases the input level by a set amount. The amount of attenuation is usually in 10dB or 20dB increments.

automation. A system that memorizes and then plays back the position of all faders and mutes on a console. In a DAW, the automation can also record and play back other parameters, including sends, returns, panning, and plug-in parameters.

B-section (also known as a pre-chorus). A section of a song between the verse and chorus sections. Not found in every song.

bandwidth. The number of frequencies that a device will pass before the signal degrades. A human being can supposedly hear from 20Hz to 20kHz, so the bandwidth of the human ear is 20Hz to 20kHz.

basic track. Recording the rhythm section for a record, which could be only the drums but could also include all the instruments of the band, depending upon the project.

bass management. A circuit that utilizes the subwoofer in an immersive audio playback system to provide bass extension for the five main speakers. The bass manager steers all frequencies below approximately 100 Hz into the subwoofer along with the LFE source signal. See LFE.

bass redirection. Another term for bass management.

big ears. The ability to be very aware of the sonic and musical details during recording or playback. The ability to rapidly dissect a track in terms of arrangement.

bit rate. The transmission rate of a digital signal.

bottom. Bass frequencies, the lower end of the audio spectrum. See also low end.

bottom end. See bottom.

bpm. Beats per minute. The measure of tempo.

Breakdown: A section of a song where the arrangement goes for being very full to very sparse.

brick wall. A limiter employing digital “look-ahead” technology that is so efficient that the signal will never exceed a certain predetermined level, so there can be no digital “overs.”

buss. A signal pathway.

butt cut. Sometimes known as a straight cut, a butt cut is an audio edit with no fade.

chamber (reverb). A method of creating artificial reverberation using a tiled room in which a speaker and several microphones are placed.

chatter. When a gate rapidly turns on and off due to fluctuating signal dynamics.

chorus. A type of signal processor where a detuned copy is mixed with the original signal to create a fatter sound.

clean. A signal with no or barely noticeable distortion.

clip. To overload and cause distortion.

clipping. When an audio signal begins to distort because a section of the signal path is overloaded, the top of the waveform becomes “clipped” off and begins to look square instead of rounded. This usually results in some type of distortion, which can be either soft and barely noticeable or horribly crunchy-sounding.

codec. An acronym for encoder/decoder.

color. To affect the timbral qualities of a sound.

comb filter. A distortion produced by combining an electronic or acoustic signal with a delayed copy of itself. The result is peaks and dips introduced into the frequency response.

compression. Signal processing that controls and evens out the dynamics of a sound.

compressor. A signal-processing device used to compress audio dynamics.

competitive level. A mix level that is as loud as your competitor's mix.

cut. To decrease, attenuate, or make less.

DAC. Digital-to-analog convertor. The device that converts the signal from the digital domain to the analog domain.

data compression. An algorithm that selectively eliminates bits from a digital stream to make it more efficient for storage and transmission.

DAW. A digital audio workstation. A computer with the appropriate hardware and software needed to digitize and edit audio.

dB. Stands for decibel, which is a unit of measurement of sound level or loudness. The smallest change in sound level that an average human can hear is 1dB.

decay. The time it takes for a signal to fall below audibility.

delay. A type of signal processor that produces distinct repeats (echoes) of a signal.

desk. A British name for a recording console.

DI. Direct inject, an impedance-matching device for guitar or bass that allows the instrument to be connected directly to recording console or DAW.

direct. To “go direct” means to bypass a microphone and connect the guitar, bass, or keyboard directly into a recording device.

direct box. See DI.

digital domain. When a signal source is converted into a series of electronic pulses represented by 1s and 0s, the signal is then in the digital domain.

digital over. The point beyond 0 on a digital processor level meter where the red Over indicator lights, resulting in a digital overload and distortion.

divergence. A parameter of surround panning that allows you to increase the level to channels other than the one panned to.

Dolby Atmos. A surround sound technology developed by Dolby Laboratories. It expands on existing surround sound systems by adding

height channels, allowing sounds to be interpreted as three-dimensional objects.

double. To play or sing a track a second time. The inconsistencies between both tracks make the part sound bigger.

double time. When one or more instrument plays the song at twice the tempo.

downmix. When a multichannel immersive mix is electronically interpreted into a playback format with fewer speakers.

dynamics (audio). Audio processors that control the dynamic range of an audio signal. These include compressors, limiters, gates, levelers, de-essers and clippers.

dynamics (music). The volume execution when an instrument is played. Songs that vary in dynamics are found to be expressive and interesting.

dynamic range. A ratio that describes the difference between the loudest and the quietest audio. The higher the number equals a greater dynamic range.

dubbing mixer. A film mixer who works on a dubbing stage, which is a film theater with an audio console placed in the middle.

edgy. A sound with an abundance of midrange frequencies.

element. A component or ingredient of the mix.

envelope. The attack, sustain, and release of a sound.

equalizer. A tone control that can vary in sophistication from very simple to very complex. See parametric equalizer.

equalization. Adjustment of the frequency spectrum to even out or alter tonal imbalances.

exciter. An audio effects processor that uses phase manipulation and harmonic distortion to produce high-frequency enhancement of a signal.

feedback. When part of the output signal is fed back into the input.

feel. The groove of a song and how it feels to play or listen to it.

flanging. The process of mixing a copy of the signal back with itself, but gradually and randomly slowing the copy down, causing the sound to

“whoosh” as if it were in a wind tunnel. This was originally done by holding a finger against a tape flange (the metal part that holds the magnetic tape on the reel), hence the name.

flip the phase. Selecting the phase switch on a console, preamp, or DAW channel in order to find the setting with the greatest bass response.

football. A musical whole note. Long sustaining chords.

FS. Full scale. A digital peak meter that reads at 0dB shows the full scale of the meter. The maximum amplitude of a digital system.

gain. The amount that a sound is boosted.

gain reduction. The amount of compression or limiting.

gain staging. Setting the gain of each stage in the signal path so that the audio level from one stage doesn’t overload the next one in line.

grid. The spaced lines on a DAW timeline that represents each beat and sub-beat.

groove. The pulse of the song and how the instruments dynamically breathe with it. Or, the part of a vinyl record that contains the mechanical information that is transferred to electronic info by the stylus.

Haas Effect. A psychoacoustic effect where any delay signal below 40 milliseconds is indistinguishable from the source event. In other words, instead of hearing the sound and then a delay (two events), you hear both the source and the delay together as a single event.

headroom. The amount of dynamic range between the normal operating level and the maximum level, which is usually the onset of clipping.

Hz. An abbreviation for hertz, which is the measurement unit of audio frequency, meaning the number of cycles per second. High numbers represent high frequency sounds, and low numbers represent low frequency sounds.

high end. The high frequency response of a device.

high-pass filter. An electronic device that allows the high frequencies to pass while attenuating the low frequencies. Used to eliminate low-frequency artifacts like hum and rumble. The frequency point where it cuts off can be fixed, switchable, or variable.

hook. A catchy phrase either played or sung.

hyper-compression. Too much buss compression or limiting during mixing or mastering in an effort to make the recording louder results in what's known as hyper-compression, a condition that essentially leaves no dynamics and makes the track sound lifeless.

I/O. The input/output of a device.

immersive audio. Multi-dimensional sound that completely envelops the listener because of speakers placed around the listening environment as well as overhead.

input pad. An electronic circuit that attenuates the signal, usually 10 or 20dB. See also attenuation pad.

in the box. Mixing with the software console inside a DAW application on a computer instead of using a hardware console.

iso booth. Isolation booth. An isolated section of the studio designed to eliminate outside sound from coming into the booth or sound leaking out.

intonation. The accuracy of tuning anywhere along the neck of a stringed instrument like a guitar or bass. Also applies to brass, woodwinds, and piano.

knee. The speed at which a compressor will turn on once it reaches threshold. A soft knee turns on gradually and is less audible than a hard knee.

kHz. One kHz equals 1,000 hertz (example: 4kHz = 4,000 Hz).

lacquer. The vinyl master, which is a single-sided 14” disc made of aluminum substrate covered with a soft cellulose nitrate. A separate lacquer is required for each side of a vinyl record. Since the lacquer can never be played, a ref or acetate is made to check the disc.

latency. Latency is a measure of the time it takes (in milliseconds) for your audio signal to pass through your system during the recording process. This delay is caused by the time it takes for your computer to receive, understand, process, and send the signal back to your outputs.

leakage. Sound from a distant instrument “bleeding” into a mic pointed at another instrument. Acoustic spill from a sound source other than the one intended for pickup.

Leslie. A speaker cabinet that features rotating speakers primarily used with organs.

LFE. Low-frequency effects channel. This is a special channel of 30Hz to 120Hz information primarily intended for special effects, such as explosions in movies. The LFE has an additional 10dB of headroom to accommodate the required sound pressure level of the low frequencies.

limiter. A signal-processing device used to constrict or reduce audio dynamics, reducing the loudest peaks in volume.

look-ahead. In a digital processor, look-ahead delays the audio signal a small amount (about two milliseconds) so that the processor can anticipate the transients in such a way that it catches the peak before it gets by.

loop. A small audio file, usually only four or eight beats (or measures) long that's edited in a way so that it can seamlessly repeat.

low-pass filter (LPF). A electronic frequency filter that allows only the low frequencies to pass while attenuating the high frequencies. The frequency point where it cuts off is usually either switchable or variable. Sometimes called "high cut."

low end. The lower end of the audio spectrum, or bass frequencies usually below 200Hz.

make-up gain. A control on a compressor/limiter that applies additional gain to the signal. This is helpful because the signal is automatically decreased when the compressor is working. Make-up gain "makes up" the gain and brings it back to where it was prior to being compressed and beyond.

master. A final version of a recording that is destined for distribution.

mastering. The process of turning a collection of songs into a record by making them sound like they belong together in tone, volume, and timing (spacing between songs on an album). A more modern definition is that mastering is the process of fine-tuning the level, frequency balance, and metadata of a track in preparation for distribution.

metadata. Data that describes the primary data. For instance, metadata can be data about an audio file that indicates the date recorded, sample rate, resolution, artist, record label, publisher, and so on.

midrange. Middle frequencies starting from around 250Hz and going up to 4,000Hz.

mix buss. The audio signal path network that mixes all of the individual channels together for your final mix.

modeling. Developing a software algorithm that is an electronic representation of the sound of a hardware audio device down to its smallest behaviors and nuances.

modulation. Using a second signal to modify the first. For example, a chorus uses a very low-frequency signal to modulate the audio signal and produce the effect.

mono. Short for monaural, or single audio playback channel.

monaural. A mix that contains a single channel and usually comes from only one speaker.

MP3. A data-compression format used to make audio files smaller in size.

muddy. Non-distinct because of excessive low frequencies.

mult. A section of the patchbay that enables patching to multiple inputs or outputs.

multi-band compression. A compressor that is able to individually compress different frequency bands as a means of providing more control over the compression process.

mute. An on/off switch. To mute something means to turn it off.

outboard gear. Hardware devices such as compressors, reverbs, and effects boxes that are not built into a console and usually reside in an equipment rack in the control room.

outro. The section of a song after the last chorus until the end of the song.

overs. Digital overs occur when the level is so high that it attempts to go beyond 0dB Full Scale on a typical digital level meter found in just about all digital equipment. A red Overload indicator usually will light, accompanied by the crunchy, distorted sound of waveform clipping.

overdub. To record a new track while listening to previously recorded tracks.

overtone. The harmonic part of a sound that gives it its character and uniqueness.

out of phase. The polarity of two channels (it could be the left and right channels of a stereo program) are reversed, thereby causing the center of the program (such as the vocal) to diminish in level. Electronically, when one cable is wired backwards from all the others.

pan. Short for panorama. Indicates the position of an instrument within the stereo soundfield.

panning. Moving a sound across the stereo soundfield.

parametric equalizer. A tone control where the gain, frequency, and bandwidth are all variable.

peaks. A portion of an audio signal that’s temporarily much higher in level than the rest of the signal.

phase. The relationship between two separate sound signals when combined into one.

phantom image. In a stereo system, if the left and right channels have an equal loud audio signal, the resultant sound appears to come from in between them.

phase shift. The process during which some frequencies are slowed down ever so slightly as they pass through a device. This is usually exaggerated by excessive use of equalization and may be undesirable.

phase meter. A dedicated meter that displays the relative phase of a stereo signal.

plate (reverb). A method used to create artificial reverberation using a large steel plate with a speaker and several transducers connected to it.

playlist. In radio, a list of the music that station will broadcast. In streaming music, a curated list of suggested songs.

plug-in. An add-on to a computer application that adds functionality to it. EQ, compression, modulation, and reverb are examples of DAW plug-ins.

point. The frequencies between 2k and 5kHz that cause a sound to be more distinct.

power chords. Long, sustaining, distorted guitar chords.

power trio. A three-piece band consisting of guitar, bass and drums that generally play rock or metal.

pre-chorus. See B-section.

predelay. A variable length of time before the onset of reverberation. Predelay is often used to separate the source from the reverberation so the source can be heard more clearly.

preroll. A short length of time before recording begins or a song section arrives.

presence. Accentuated high frequencies (anywhere from 5k to 10kHz).

producer. The equivalent of a movie director, the producer has the ability to craft the songs of an artist or band technically, sonically, and musically.

proximity effect. The inherent low-frequency boost that occurs with a directional microphone as it gets closer to the signal source.

Pultec. An equalizer made during the 1950s and 60s by Western Electric that is highly prized today for its smooth sound.

pumping. When the level of a mix increases and then decreases noticeably. Pumping is caused by the improper setting of the attack and release times on a compressor.

punchy. A description for a quality of sound that infers good reproduction of dynamics with a strong impact. The term sometimes means emphasis in the 200Hz and 5kHz areas.

Q. The bandwidth of a filter or equalizer that may be fixed or variable. Q stands for filter "quality."

range. On a gate or expander, a control that adjusts the amount of attenuation that will occur to the signal when the gate is closed.

ratio. A control on a compressor/limiter that determines how much compression or limiting will occur when the signal exceeds the threshold.

recall. A system that memorizes the position of all pots and switches on a console. On older analog consoles, the engineer must still physically reset

the pots and switches back to their previous positions as indicated on a video monitor.

record. A generic term for the distribution format of a recording. Regardless of whether it’s a CD, vinyl, or a digital file, it is still known as a record.

release. The last part of a sound. On a compressor/limiter, a control that affects how that device will respond to the release of a sound. Also, making a record available for distribution.

resonance. See resonant frequency.

resonant frequency. A particular frequency or band of frequencies that is accentuated, usually due to some extraneous acoustic, electronic, or mechanical factor.

return. An input on a recording console especially dedicated for effects devices, such as reverbs and delays. The return inputs are usually not as sophisticated as normal channel inputs on a console. In most DAWs, the returns are called Aux channels.

reverb. A type of signal processor that reproduces the spatial sound of an environment (such as the sound of a closet or locker room or inside an oil tanker).

rhythm section. The instruments in a band that give the song its pulse, usually the bass and drums.

RMS meter. A meter that reads the average level of a signal.

roll off. To attenuate either end of the frequency spectrum.

scratch vocal. A temporary vocal recorded during basic tracking with the intention of replacing it with a more suitable one later.

shelving curve. A type of equalizer circuit used to boost or cut a signal above or below a specified frequency; looks flat like a shelf when graphed. Usually the high- and low-band equalizers built into many mixing boards are the shelving type.

sibilance. A short burst of high frequencies centering anywhere in a vocal’s 3kHz to 10kHz range, resulting in the “S” sounds being overemphasized.

sidechain. A separate signal path to and from the control element of a dynamics device.

signal path. The electronic or digital pathway through circuitry and processors that the audio signal must pass through.

song form. The multiple sections that make up a song. Most songs have a combination of intro, verse, pre-chorus, chorus, bridge and outro.

soundfield. The listening area containing mostly direct sound from the monitor speakers.

source. An original master that is not a copy or a clone.

spectrum. The complete audible range of audio signals.

SPL. Sound-pressure level. The volume level of a sound to the human ear.

stage. In an analog console, a block of circuitry that performs a console function, such as EQ or panning. In a digital or software console, a digital block that performs a console function.

standing waves. An acoustic property of a room where certain frequencies reflect off the walls, floor, or ceiling that will either boost the signal or attenuate it, depending upon where in the room you're standing.

stem. An instrument group of tracks that make up a full mix. Stems are typically divided into drum stems, bass stem, vocal stem and instruments stem, although they may get further defined, such as background vocal

stem, keyboards stem, etc. Each stem contains all of the processing and effects added during the mix.

sympathetic vibration. Vibrations, buzzes, and rattles or notes that occur in areas of an instrument or instruments other than the one that was struck.

subgroup. A separate submixer that sums the assigned channels together and then sends that mix to the master mix buss.

sub. Short for subwoofer.

subwoofer. A low-frequency speaker with a frequency response from about 30Hz to as high as 120Hz.

synchronization. When two devices—usually storage devices, such as tape machines, DAWs, or sequencers—are locked together with respect to time.

tape slap. A method to create a delay effect using a tape machine. A slap is usually 100 to 200ms.

tempo. The rate of speed at which a song is played.

tension and release. Building a listener's expectations and then relaxing them, such as dissonance to harmony or loud to quiet.

threshold. The point at which an effect takes place. On a compressor/limiter, for instance, the threshold control adjusts the signal level at which compression will take place.

track sharing. When a single track shares more than one instrument. For instance, when a percussion part is recorded on a guitar solo track in places that the guitar has not been recorded.

timed delay. A delay where the repeats are timed to pulse along with the pulse of the song.

top-end. See high-end.

track. A term sometimes used to mean a song. In recording, a separate musical performance that is recorded to a separate piece of the DAW timeline.

transient. A very short-duration signal.

TV mix. A mix without the lead vocals so the artist can sing live to the backing tracks during a television appearance.

vamp. To repeat a short passage of music.

About Bobby Owsinski

Producer/engineer Bobby Owsinski is one of the best selling authors in the music industry with 25 books that are now staples in audio recording, music, and music business programs in colleges around the world, These include the The Music Mixing Workbook, Social Media Promotion For Musicians, The Mixing Engineer's Handbook and more. He's also a contributor to Forbes writing on the new music business, his popular blogs and Inner Circle podcast have won numerous awards, and he's appeared on CNN and ABC News as a music branding and audio expert.

Visit Bobby's music production blog at bobbyowsinskiblog.com, his Music 3.0 music industry blog at music3point0.com, his podcast at bobbyoinnercircle.com, his online courses at bobbyowsinskicourses.com, and his website at bobbyowsinski.com.

Bobby Owsinski Bibliography

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